



2N[®] Omega Lite

Ethernet Module



User Manual

Version

1.4.0

www.2n.cz

The 2N TELEKOMUNIKACE a.s. joint-stock company is a Czech manufacturer and supplier of telecommunications equipment.



The product family developed by 2N TELEKOMUNIKACE a.s. includes GSM gateways, private branch exchanges (PBX), and door and lift communicators. 2N TELEKOMUNIKACE a.s. has been ranked among the Czech top companies for years and represented a symbol of stability and prosperity on the telecommunications market for almost two decades. At present, we export our products into over 120 countries worldwide and have exclusive distributors on all continents.



2N[®] is a registered trademark of 2N TELEKOMUNIKACE a.s.. Any product and/or other names mentioned herein are registered trademarks and/or trademarks or brands protected by law.



Declaration of Conformity

2N TELEKOMUNIKACE a.s. hereby declares that the 2N[®] Omega Lite product complies with all basic requirements and other relevant provisions of the 1999/5/EC directive. For the full wording of the Declaration of Conformity see the CD-ROM enclosed and at www.2n.cz.



The 2N TELEKOMUNIKACE company is a holder of the ISO 9001:2000 certificate. All development, production and distribution processes of the company are managed by this standard and guarantee a high quality and advanced technical level of and a professional approach to all of our products.

Dear customer,

*let us congratulate you on having purchased the **2N® OMEGA Lite** system. This new product has been developed and produced in order to provide the maximum utility value, quality and reliability to the user.*

We hope you will be fully satisfied with 2N® OMEGA Lite for a long time.

CONTENTS

1. Packing List.....	6
2. Ethernet Module.....	7
2.1. Ethernet Module	8
3. Ethernet Module Installation	9
4. Ethernet Module Configuration.....	17
4.1. USB Cable Connection	18
4.1.1. Submenus.....	19
4.1.2. Commands and Values	19
4.1.3. About Menus	20
4.1.4. DTR Signal Monitoring	20
4.2. Web Browser Access.....	22
4.2.1. Overview of Group Tags and Menus	24
4.3. Network Parameter Setting.....	27
4.3.1. Serial Console Setting.....	27
4.3.2. Web Browser Setting	28
4.4. Administration	30
4.4.1. General Module Parameters.....	30
4.4.2. User Accounts.....	31
4.4.3. Right Groups.....	35
4.4.4. Ring groups.....	36
4.4.5. Emergency Change of Administrator Password	38
4.4.6. SIP / H.323 Switching	39
4.5. Communication Line Setting.....	39
4.5.1. SIP	39
4.5.2. SIP Line Parameters.....	40
4.5.3. H.323	42
4.5.4. H.323 Line Parameters	43
4.6. Least Cost Router	46
4.6.1. Routing Rules	47
4.6.2. Add Rule	47
4.6.3. Modify Rule.....	49
4.6.4. Remove Rule	51
4.6.5. Change Sequence.....	51
4.6.6. Disable/Enable Rule	51
4.7. LCR Routes	51
4.7.1. List of Routes.....	52
4.7.2. Add Route	52
4.7.3. Modify Route.....	54
4.7.4. Remove Route	54
4.8. LCR Number Normalising.....	54
4.9. LCR Barred Numbers	56
4.10. LCR Testing	56
4.11. SIP Proxy.....	58
4.11.1. Overview of Registrations	61
4.11.2. Switching ON/OFF	61
4.12. Mobility Extension	62
4.12.1. User Settings	62
4.12.2. DISA Line for ME	63
4.12.3. Mobility Extension.....	64
4.12.4. LCR – Least Cost Router	66

4.12.5.	PBX Settings	73
4.13.	PPPoE	76
4.13.1.	Main configuration.....	76
4.13.2.	DHCP server settings	78
4.13.3.	DNS proxy	80
4.14.	Licence File.....	83
4.15.	Firmware Updating	84
4.16.	Configuration Downloading.....	85
4.17.	Configuration Uploading	86
5.	<i>Ethernet Module Parameter Setting by PBX Configuring Tool</i>	87
5.1.	2N® OMEGA PBX Firmware Upgrade	87
5.2.	Hardware Configuration	87
5.3.	VoIP Line Setting	89
5.4.	Outgoing Calling	89
5.5.	Incoming Ringing	90
6.	<i>2N® XAPI Server.....</i>	91
6.1.	2N® XAPI Server Console	91
6.1.1.	Installation	92
6.1.2.	Communication Setting	96
6.1.3.	Connection Establishing	98
6.1.4.	Connection Closing.....	98
6.1.5.	PBX Module	99
6.1.6.	User Module	102
6.1.7.	Licence Module	105
6.1.8.	Database Module	108
6.1.9.	XML Module	110

1. Packing List

Item	Pieces
Ethernet module	1
USB cable	1
PC LAN cable	1
Connector	2
M3x30 spacer	2
Warranty Certificate	1
Compliance Certificate	1
CD	1

2. Ethernet Module

The Ethernet module is designed exclusively for the **2N® OMEGA Lite** PBXs to provide remote control, connection of external applications (PBX Assistant, SMS server) and LAN and Internet voice services.

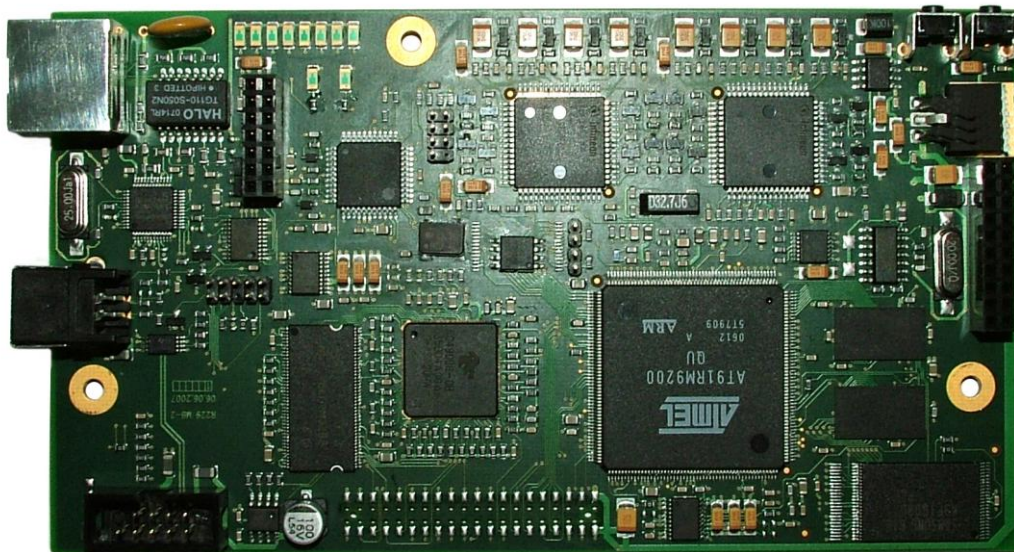
The Ethernet module is installed to the left of the power supply unit, above the main board. To connect the module, use two connecting cables (insert them on the J15/J26 main board connectors) and two spacers (all are included in the delivery). Fit the Ethernet module to the spacers using the two M3 screws included.

Remove the two J8/J9 jumpers on the PBX main board to make the system work properly.

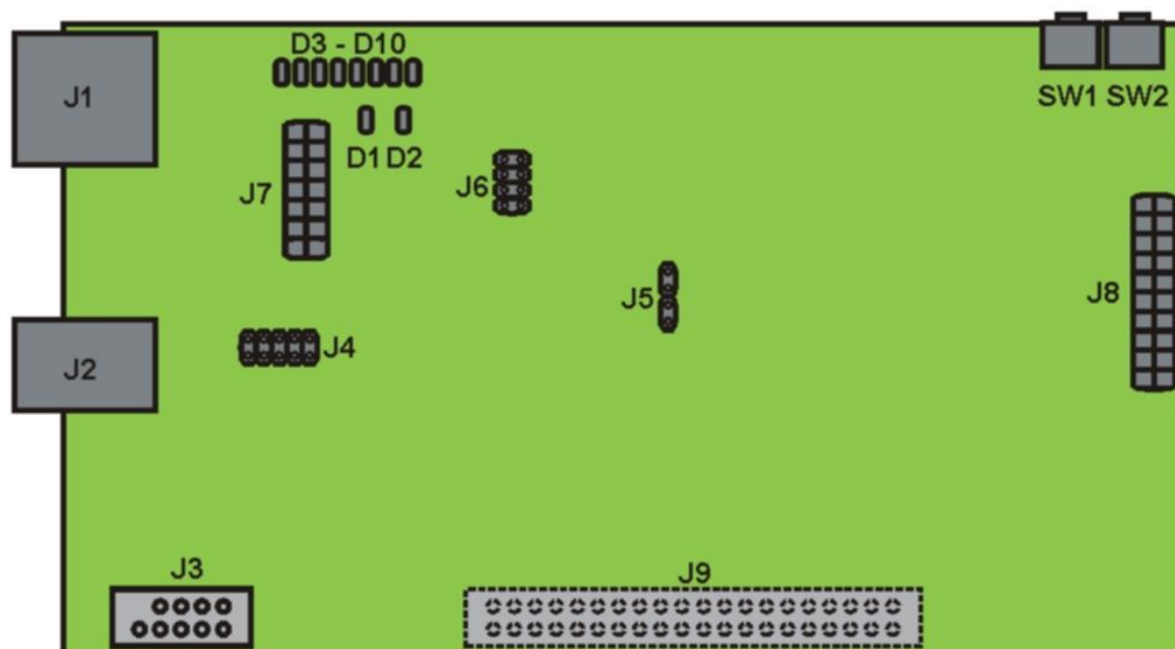
Use the USB cable and HyperTerminal for the initial settings of the IP address and network mask, and the available web configuration for further settings.

To connect the system into the 10/100BASE-T (Twisted Pair Ethernet) computer network, use a standard straight cable terminated with RJ 45 connectors.

See the Ethernet Module User Manual on the enclosed CD for detailed installation information.



2.1. Ethernet Module



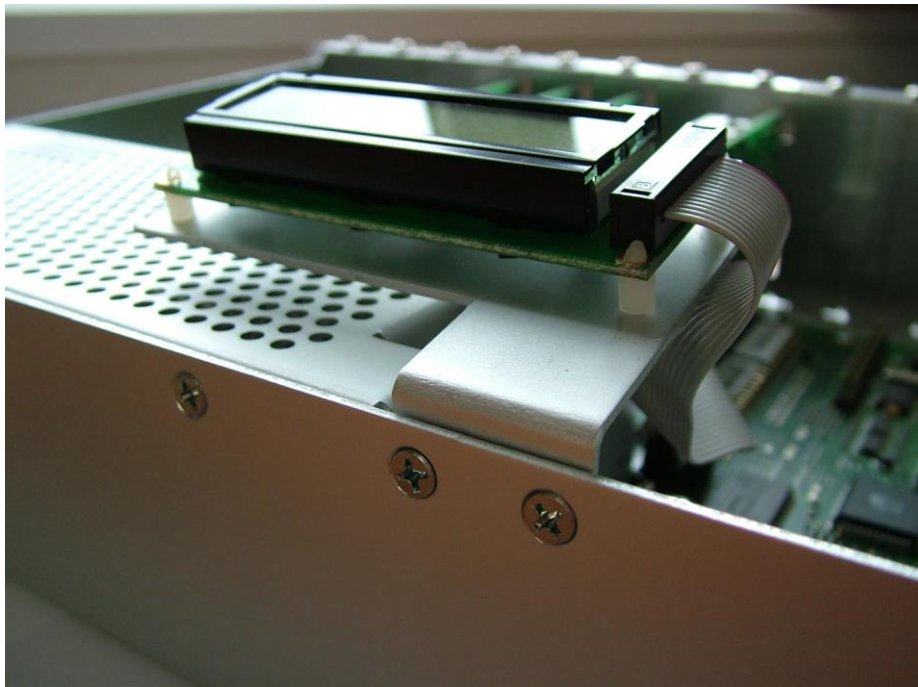
The following is located on the Ethernet module:

- J1 - RJ45 connector - 8/8 for LAN connection;
- J2 - USB connector for IP configuration;
- J4, J8 - serial connector for interconnection with the main board;
- J3 - connector for an external USB device adapter (UMTS/3G module, e.g.);
- J5-J7 - connector for services and maintenance;
- J9 - connector for an extender board;
- SW2 - RESET button - push the button to reset the CPU board;
- SW1 - RESET button - push the button to reset the default values (keep the button pushed for approximately 20s after the PBX power on until LINUX starts);
- D1 - green LED – operation status indication (module ready for operation);
- D2 - red LED – operation status indication (module inactive);
- D3-D10 - yellow LED – speech channels busy indicators.

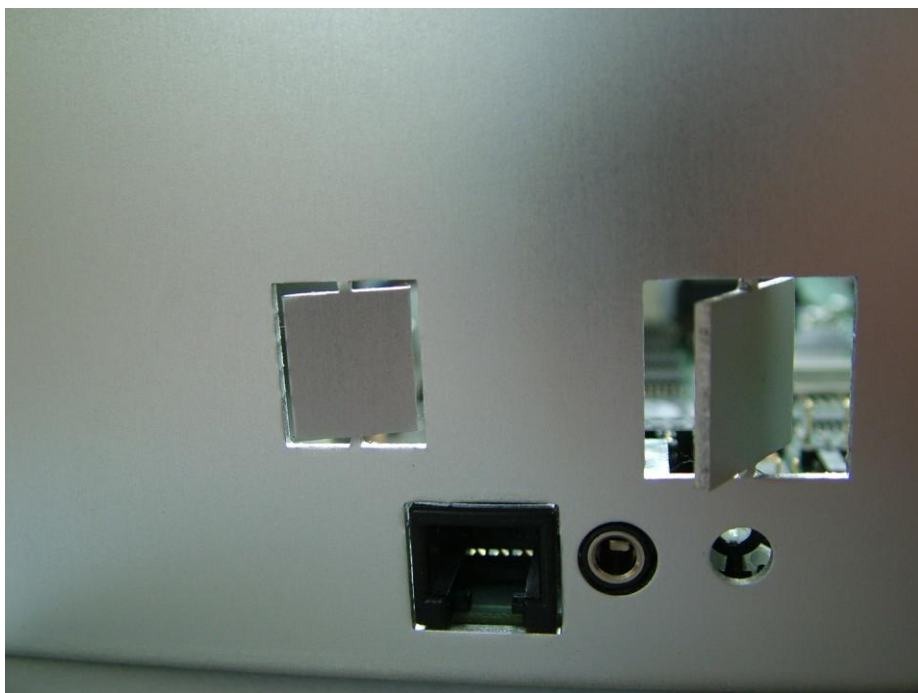
* under development

3. Ethernet Module Installation

Unscrew the 2 screws on the chassis to remove the display holder.



Remove (by turning) the blind flanges on the left-hand side of the chassis to make space for J1 and J2 connectors.

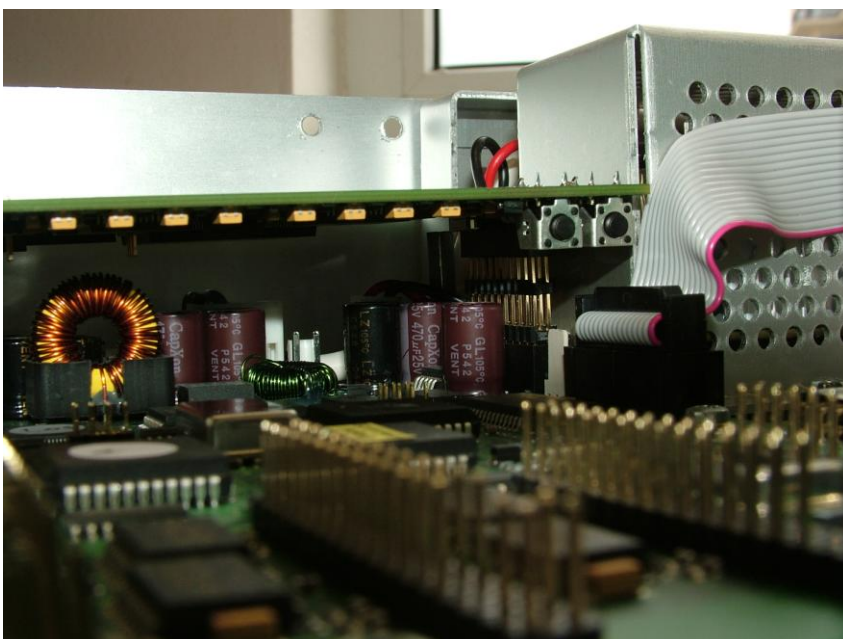


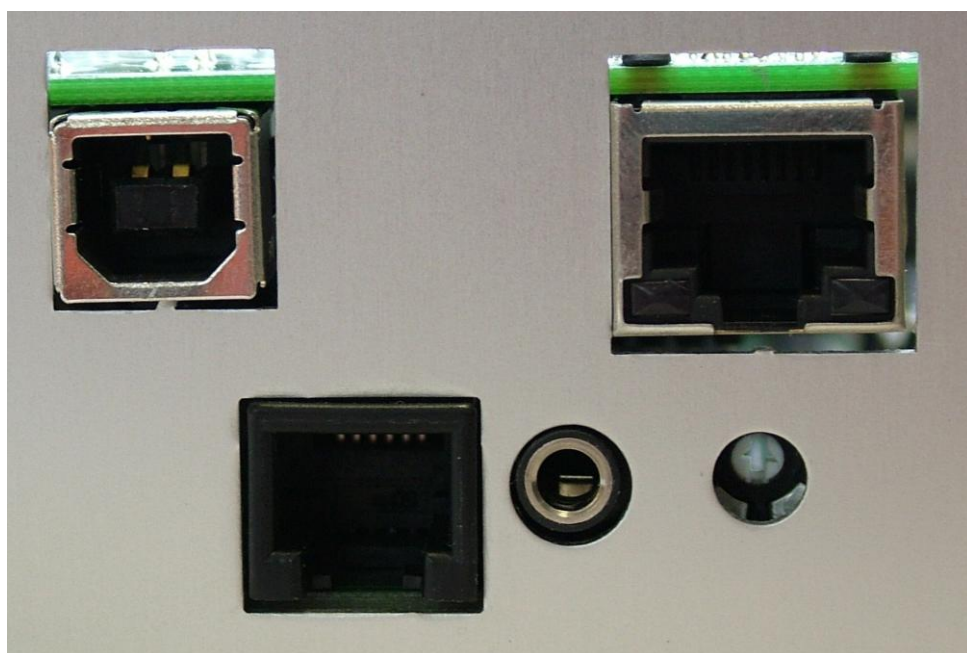
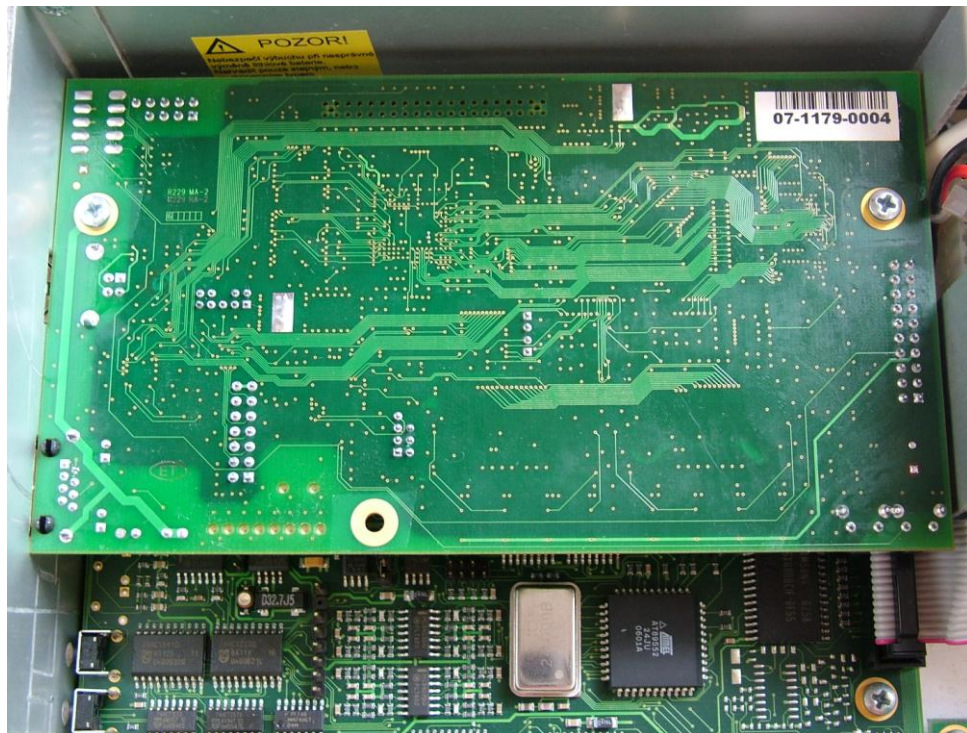


Place the module onto the connectors and fix it with the two screws removed earlier.



Place the module onto the connectors carefully to match the pin positions.

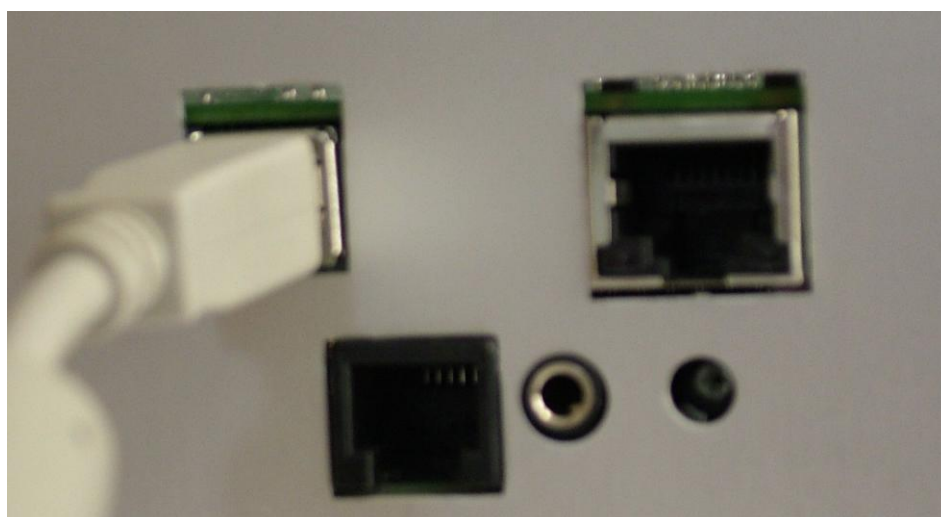
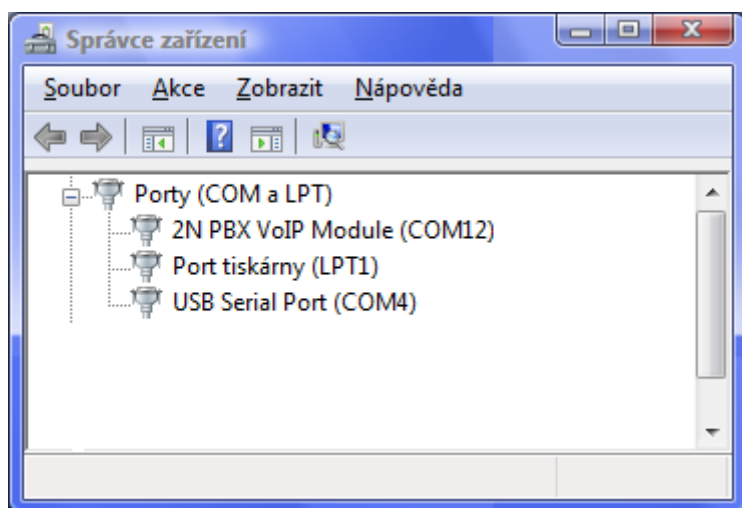




Connect the USB cable inserted to the J2 connector to make connection with your PC.



While connecting the system for the first time, remember to install the USB-COM driver (refer to the “data/OMEGA Lite/Cz/Software/VoIP/DriverUSB/” CD attached). A new COM PORT is added after installation. After that you may establish the HyperTerminal connection and perform the necessary settings.

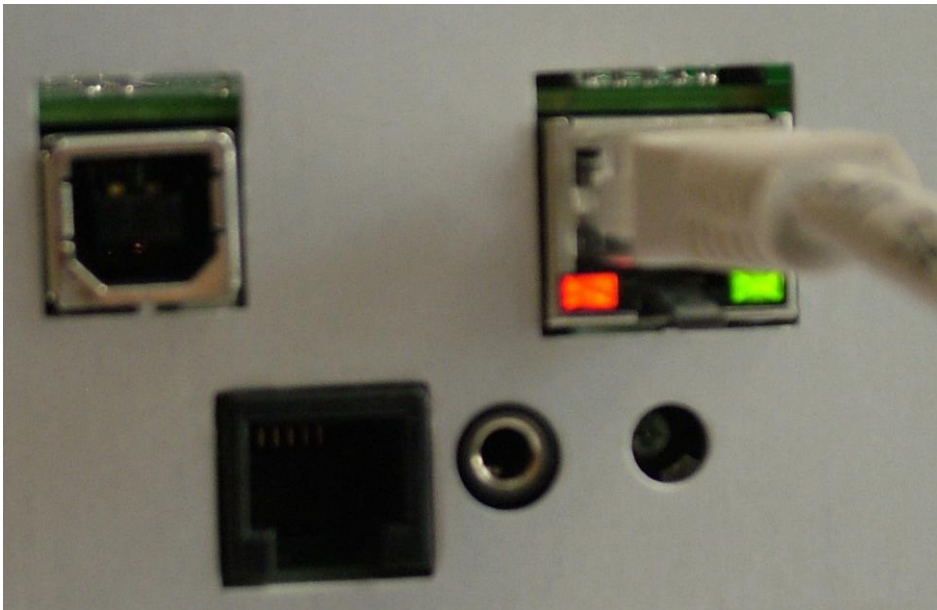


Used for network parameter setting (IP address, network mask, etc.).
After the network parameters have been set up you may disconnect the cable.



Always reset the module using the SW1 button after disconnecting the USB cable.

Connect a standard straight Ethernet cable into the J1 connector.



This cable connects the Ethernet module with your Ethernet network.
After turning the PBX on you may start configuring the Ethernet module.

4. Ethernet Module Configuration

The 2N[®] OMEGA PBX Ethernet module has the following factory settings:

Serial console communication port:

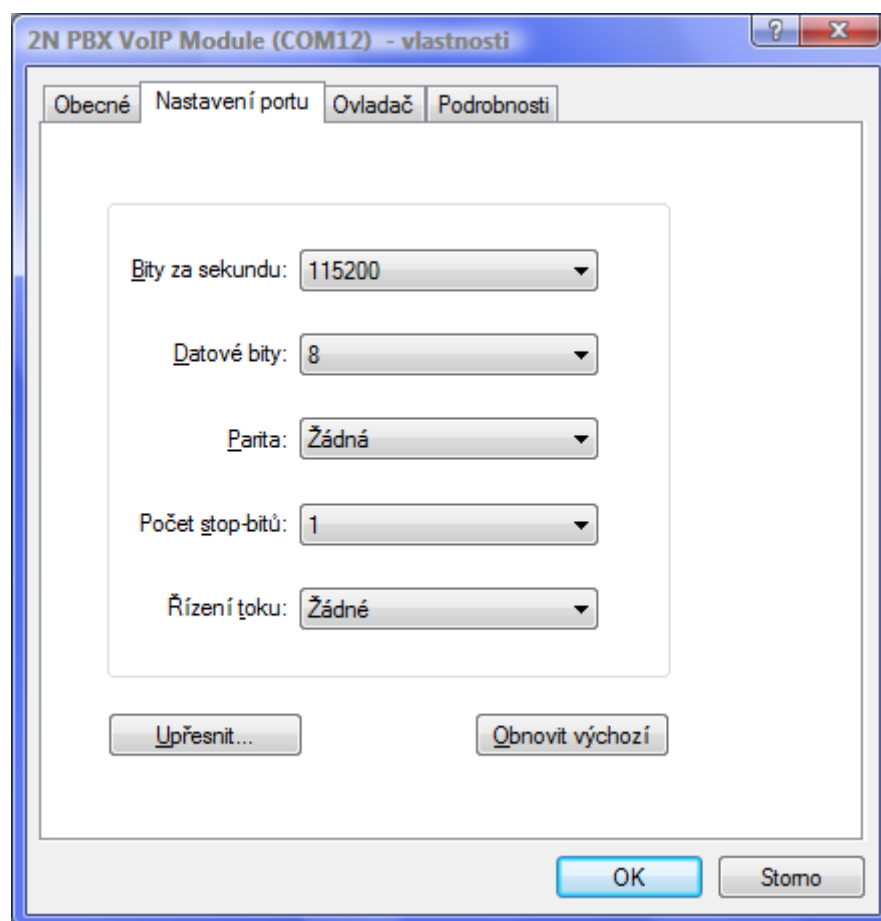
Bits per second: 115200

Data bits: 8

Parity: none

Flow control: none

Use the same values for the COM port to be created.



Network parameters:

IP address: 192.168.1.1

Network mask: 255.255.255.0

The module may be configured in two ways:

1) Connect the module to the PC using a USB cable and set the basic network parameters by means of some communication software, e.g. the HyperTerminal. Then use a web browser to complete the setting.

2) Connect the module with the PC using a crossed Ethernet cable, assign any IP address from the network 192.168.1.x differing from 192.168.1.1 to the PC and assign the required IP address to the module by means of a web browser. Having changed the IP address, connect the PC and the module to the LAN in a standard way and use a web browser to complete the configuration.


4.1. USB Cable Connection

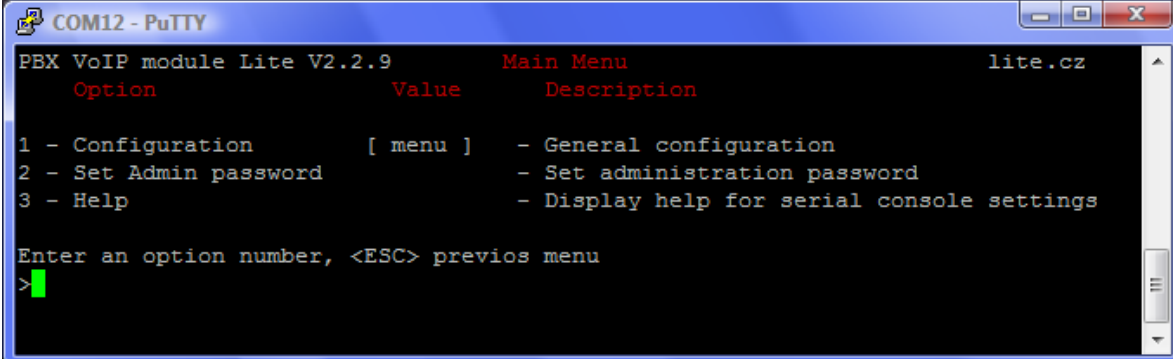
For configuration, connect the Ethernet module (J2 connector) with the PC using the available USB cable.

Install the driver from the enclosed CD to create a virtual COM port (COM4).

Launch the HyperTerminal, e.g., configure it for receiving on the COM port created for the new USB connection and set the network parameters.

The console system is arranged as a set of nested menus. Selecting an item results either in entering a submenu, executing an operation, or setting a parameter.

The main menu should appear after terminal connection if the module is connected. In some cases it is necessary to push .




```
COM12 - PuTTY
PBX VoIP module Lite V2.2.9      Main Menu      lite.cz
Option      Value      Description

1 - Configuration      [ menu ]      - General configuration
2 - Set Admin password      - Set administration password
3 - Help      - Display help for serial console settings

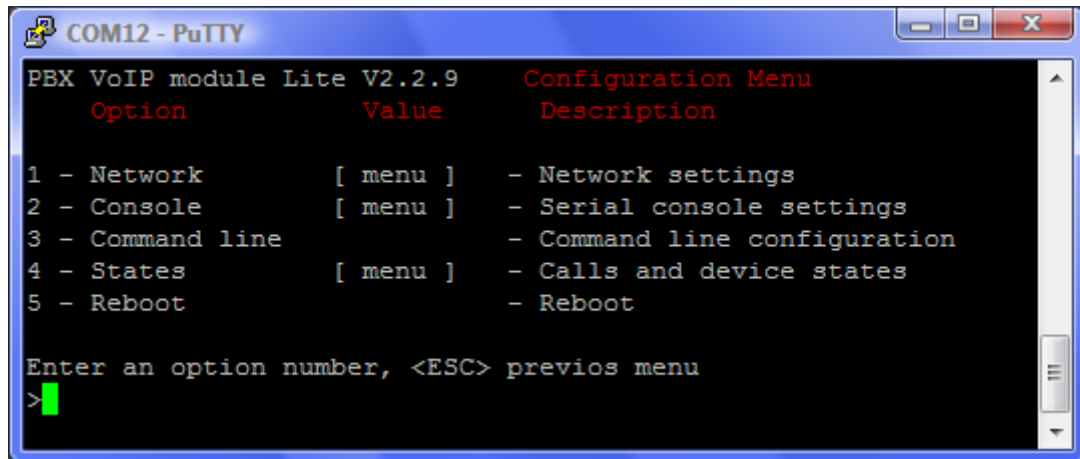
Enter an option number, <ESC> previos menu
>
```

Each menu consists of the following sections:


- **Header** - contains (from left): product name, firmware version, menu name and network name assigned to the equipment.
- **Option** - displays the numbers and names of available options.
- **Value** - if displayed, the [menu] includes a submenu. Otherwise, the current value of the option is displayed.
- **Description** - briefly describes the meaning of each option in the menu.
- **Enter an option number... >**: invites you to enter the option number.

To select a menu option, enter the option number and push  for confirmation.

4.1.1. Submenus



If you click on an item containing a submenu, the submenu will be displayed.


Select any of the submenu items or push  for return.

4.1.2. Commands and Values

Having selected an item, you may be asked to enter any of the following data:

- **Key word** - a list of one or more fixed strings. All you have to do to select a key word is enter the number of characters that distinguishes it unambiguously from the other key words.

Enter one of [ansi,color,teletype] : a

In above stated example, enter 'a', or 'c', or 't' and push  to select a word.

- **Chain** - any number of characters. The range of the available value length may be displayed.

Enter a hostname from 1 to 32 characters:
VoiceBlue

- **Integer number** - a decimal integer number. The range of the available value length may be displayed.

Enter a size between 1 and 100: 99

A hexadecimal integer number – a number entered in the hexadecimal format using the 0÷9 and a÷f or A÷F characters.

Enter a hex number between 1h and ffh: 1a

- **Network address** - up to 12 hexadecimal digits of the physical address. Zeros at the beginning may be omitted.

Enter the remote network address: 50C229C4E2

- **IP address** - an Internet address consisting of four numbers (0÷255) separated by dots.

Enter an IP address: 192.168.22.30

As soon as the required information is entered, the relevant operation is executed. If such operation changes the settings, new values will be displayed in the updated menu.

Some of the configuration parameters may take just one of two fixed values. By selecting such an item you reverse the value as compared to the state before selection. ON/OFF parameters are a typical example. If the value is ON, then it is switched to OFF when selected and vice versa.

Some commands perform operations that significantly influence the behaviour of entire system (restart, e.g.). Before executing them, the system usually requests confirmation.

Are you sure [y/n]:

If you answer anything else than “y” or “Y”, the command will be aborted.

To cancel this information prompt any time, push the  button.

4.1.3. About Menus ...

Main menu - is displayed whenever the terminal is started. You can get to the configuration menu, change the administrator password or display the help from the main menu.

Configuration menu - contains two submenus, Network configuration and Serial console configuration.

Network configuration - helps you set the IP address, network mask, default router, DNS addresses, network name and domain.

Serial console configuration - enables you to set the terminal type and change the default parameters of the serial link – rate, data bits/stop bits, parity and flow control type.

4.1.4. DTR Signal Monitoring

The program continuously monitors the Data Terminal Ready (DTR) signal. This signal is used for detection of connection/disconnection of the terminal to the serial console port.

The following activities are executed whenever the terminal disconnection is detected:

- Initiated operations are aborted.
- The main menu is displayed.

4.2. Web Browser Access

While the serial console interface enables you to change of the basic parameters only, the web browser gives you access to all Ethernet module parameters and services.

To establish connection with the module, enter the module IP address into the Internet address line in the web browser, e.g.:

<http://192.168.1.1>

If you have connected the system and set the module and PC IP addresses correctly, the user name and password entering prompt will be displayed.



The manufacturer supplies the Ethernet modules with an integrated user account *Admin* without the password. While entering the user name and password mind the Lower/Upper Case. Remember to change the administrator password as soon as you put your new module in operation for safety reasons.





After a successful login, the basic screen of the web application is displayed. The window is divided into four major parts.

Header - contains a pull-down menu for the user session language selection, the name of the currently logged-in user and the user logout button.

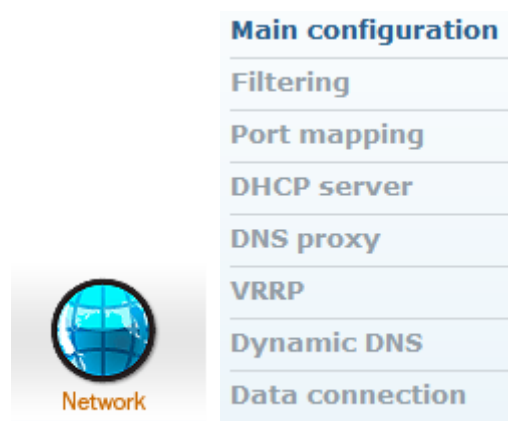
Menu tag ears - the module settings are grouped in tags. Clicking on a tag ear you access the selected setting group and get the group menu options.

Group menu - the tag settings are arranged in two-level menus. Clicking on a first-level menu item you display the second-level submenus. If you click on a first level item without submenus or a submenu, the appropriate application form will be displayed on the remaining window area.

Application form - is the key part of the module user interface. It contains specific control elements for the group menu item selected.

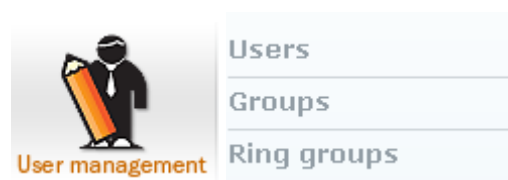
4.2.1. Overview of Group Tags and Menus

Network - here you can find all you need for your LAN connection:



- **Basic configuration** – common module settings.
- **Filtering** – port and IP address filter.
- **Port mapping** – connection to internal equipment with UMTS module.
- **DHCP server** –DHCP server settings.
- **DNS proxy** – DNS settings.
- **VRRP** – VRRP settings.
- **Dynamic DNS** –dynamic DNS settings.
- **Data connection** – UMTS module connection settings.

User management - here you can find all you need for user account administration:



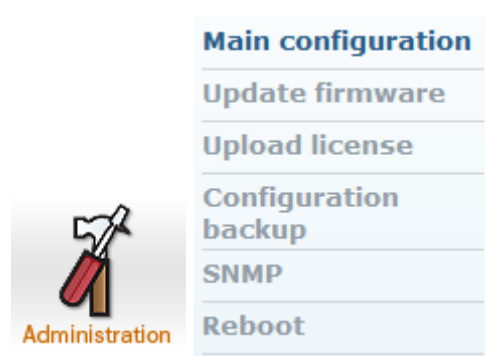
- **Users** – administration of own accounts.
- **Groups** – administration of account groups.
- **Ring groups** – administration of Ring Groups.

Telephony services - here you can find all voice service settings:



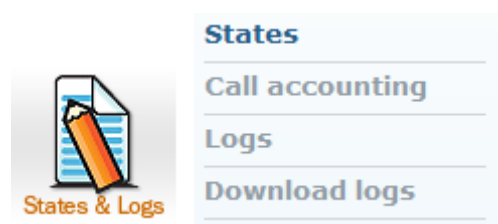
- **Devices** – common module settings.
- **Fax** – fax settings.
- **LCR** – LCR rule settings.
- **SIP proxy** – SIP Express Router settings (SIP proxy).

Administration - here you can find all basic module settings:



- **Basic configuration** – common module settings.
- **Update firmware** – module firmware updating.
- **Upload licence** – module licence updating.
- **Configuration backup** – module configuration backup.
- **SNMP** – SNMP settings.
- **Reboot** – module reset.

States & Logs - activity records are generated during module operation. For the records, their settings and current states refer to the *States & Logs* tag menu.



- **States** – information of current calls.

- **Call accounting** – call billing records and settings.
- **Logs** – firmware operation records.
- **Download logs** – operation record packing and downloading.

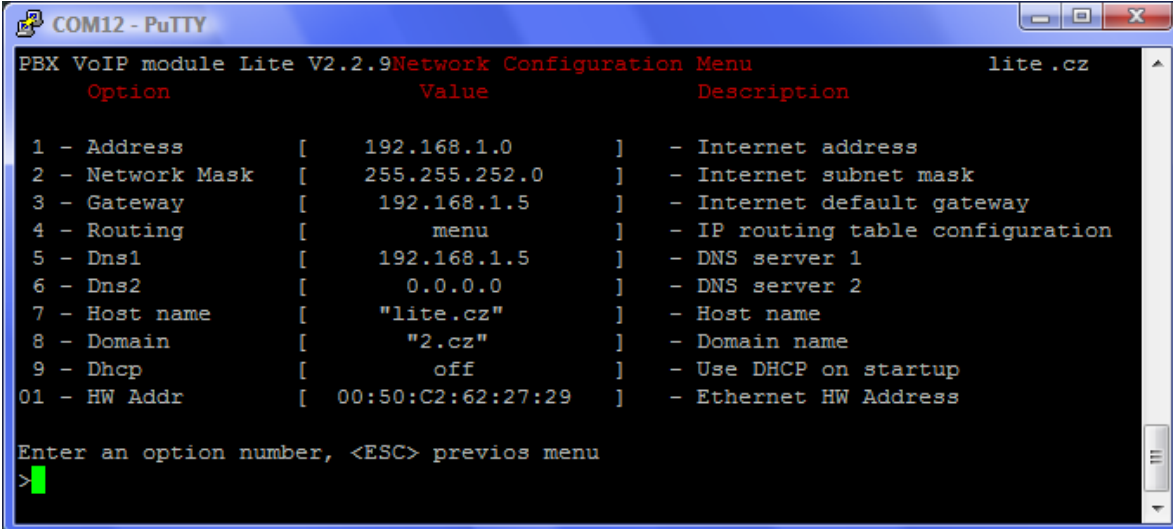
4.3. Network Parameter Setting

Remember to set the module network parameters before you start using it for calling. There are two setting options. If the gateway has factory settings or you know its IP address, you can connect a PC to it using a crossed Ethernet cable and set the appropriate parameters via a web browser.

In other cases, you will surely appreciate the serial port access via a character terminal, e.g. the HyperTerminal.

4.3.1. Serial Console Setting

Connect the module to a PC as described in the Serial console configuration section. The main menu gets displayed on the terminal. Here select the *Configuration* item and then the *Network* item and get the following menu:



```
COM12 - PuTTY
PBX VoIP module Lite V2.2.9Network Configuration Menu      lite.cz
Option              Value              Description
1 - Address         [ 192.168.1.0      ] - Internet address
2 - Network Mask    [ 255.255.252.0    ] - Internet subnet mask
3 - Gateway         [ 192.168.1.5      ] - Internet default gateway
4 - Routing         [ menu             ] - IP routing table configuration
5 - Dns1            [ 192.168.1.5      ] - DNS server 1
6 - Dns2            [ 0.0.0.0          ] - DNS server 2
7 - Host name       [ "lite.cz"        ] - Host name
8 - Domain          [ "2.cz"           ] - Domain name
9 - Dhcp            [ off              ] - Use DHCP on startup
10 - HW Addr        [ 00:50:C2:62:27:29 ] - Ethernet HW Address

Enter an option number, <ESC> previos menu
>
```

The menu contains the following items:

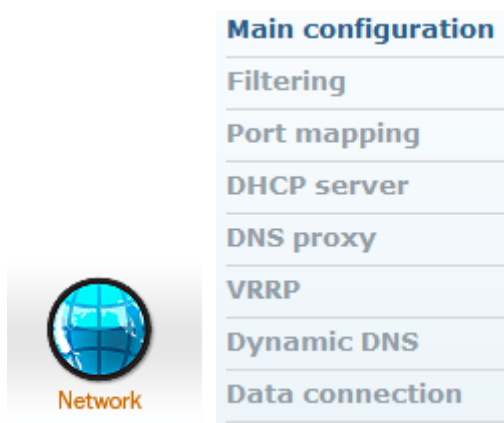
- **Address** – is the module IP address. If the dynamic address assignment by the DHCP is activated, the value may not be changed manually.
- **Network mask** – is the network mask. If the dynamic address assignment by the DHCP is activated, the value may not be changed manually.
- **Gateway** – is the default router IP address. It is used for routing data beyond the network limits. If the dynamic address assignment by DHCP is activated, the value may not be changed manually.
- **DNS 1** – first DNS IP address.

- **DNS 2** – second DNS IP address.
- **Hostname** – Ethernet module network name.
- **Domain** – domain name.
- **Location** – any text describing the module location. This value is published by means of the SNMP.
- **Contact** – information on the module administrator. This value is published by means of the SNMP.
- **DHCP** – indicates the dynamic IP address assignment. If it is activated, the module gets its network settings from the DHCP. Otherwise, the data must be entered manually.
- **Class** – device class name as included in the network-setting sending request. It helps the DHCP to distinguish various types of devices and assign them configuration parameters accordingly.

The DHCP and Address/Network mask parameters at least must be set correctly to enable the TCP/IP module communication. You are also recommended to set the address of one DNS at least.

4.3.2. Web Browser Setting

If you use a web browser for module connection and know the Administrator password, you can set the network parameters in the *Basic configuration* menu in the *Network* tag. After opening of the menu, the current network setting is displayed. To open the form for changes, click on the *Change* item located to the right, below the network parameter tables. The following form gets displayed:





The image shows a network configuration interface for the **eth0** interface. It includes a checkbox for 'Enable DHCP' and three text input fields for 'IP address', 'Network mask', and 'Gateway'. The values entered in the fields are 192.168.1.215, 255.255.252.0, and 192.168.1.5 respectively.

Field	Value
Enable DHCP:	<input type="checkbox"/>
IP address:	192.168.1.215
Network mask:	255.255.252.0
Gateway:	192.168.1.5

The form contains the following fields:

- **Enable DHCP** – indicates whether the network parameters can be obtained from the DHCP, or have to be entered manually.
- **IP address** – module IP address. If the dynamic address assignment by the DHCP is activated, the value may not be changed manually.
- **Network mask** – network mask. If the dynamic address assignment by the DHCP is activated, the value may not be changed manually.
- **Gateway** – default router IP address. It is used for routing data beyond the network limits. If the dynamic address assignment by the DHCP is activated, the value may not be changed manually.

DNS	
Hostname:	lite.2n.cz
DNS server 1:	192.168.1.5
DNS server 2:	0.0.0.0
Domain:	2n.cz

- **DNS 1** – first DNS IP address.
- **DNS 2** – second DNS IP address.
- **Domain** – domain name.

Click on the *Change* item to confirm the changes. It takes some time to set the values. Please wait for the web browser response.

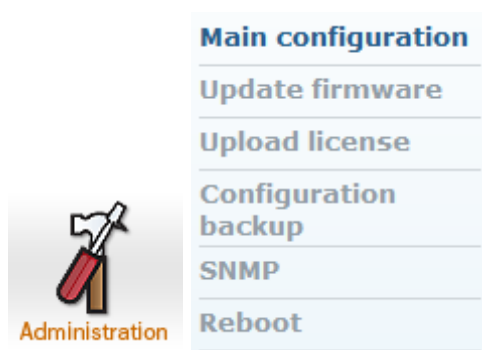
To restore the state before form opening, push the *Default* item.

4.4. Administration

This chapter describes the setting of general parameters of the 2N[®] OMEGA Lite Ethernet module and creating of user accounts and right groups.

4.4.1. General Module Parameters

Click on the *Basic configuration* item in the *Administration* tag to open the page with the general module parameters. You cannot change the parameter values in this form. To enter the change mode, click on the *Change* item.



Administration

Confirm remove: ☒

Enable SIP session progress: ☒

Advanced config of regular expressions: ☐

Default language:

Max user session time: (seconds)

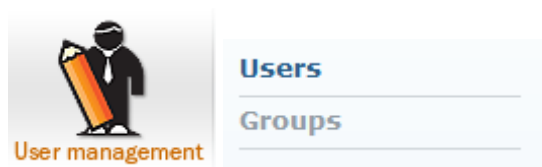
Simple login page: ☐

The form contains the following fields:

- **Confirm remove** – select this item to display the confirmation window before any deletion.
- **Enable SIP session progress** – select this option to enable the SIP stack to send the session progress messages during call establishing.
- **Advanced config of regular expressions** – here enable the use of advanced regular expressions in the LCR/Normalization section. It changes format of Add LCR Normalization menu !!!
- **Default language** – select the language for the web interface login page.
- **Maximum user session time** – set the inactivity timeout after which the web interface relation will be disconnected automatically.
- **Simple login page** – if you select this option, the login page will be displayed without the graphic mode.

4.4.2. User Accounts


To define user accounts for the web interface login or SIP telephone authentication to the integrated SIP proxy, select the *Users* menu in the *User management* tag. After selection, the list of user accounts will be displayed.




Users							
Name	Group	Language	Line number	Description			
11	PBX users	English	20	Administrator			<input type="checkbox"/>
12	PBX users	English					<input type="checkbox"/>
20	Administrators	English					<input type="checkbox"/>
Admin	Administrators	English					<input type="checkbox"/>
siphone	Administrators	English					<input type="checkbox"/>

The following items are displayed in the table columns:

- **Name** – name of the user account.
- **Group** – group of rights.
- **Language** – language to be used after the user login.
- **Line number** – number to be used for line registration.
- **Description** – any text describing the meaning of an account.

Click on the  symbol to move to the user account detail for making changes.


The  symbol is used for clearing the account. To clear an account, you can also tick off the field in the last column of the table and click on *Remove selected item*.

There is an *Add user* icon below the table. Click on it to display the user defining form.



Add user

User name:
New password:
Confirm new password:
Group:
Language:
Default application:
Rights: ☐ USERS+LINES+LCR ☐ USERS ☐ LINES ☐ LCR ☐ SMS
Rights denied: ☐ USERS+LINES+LCR ☐ USERS ☐ LINES ☐ LCR ☐ SMS
Line number:
Description:
Voicemail
Voicemail enabled: ☐
PIN:
SMS notification number:



The form contains the following fields:

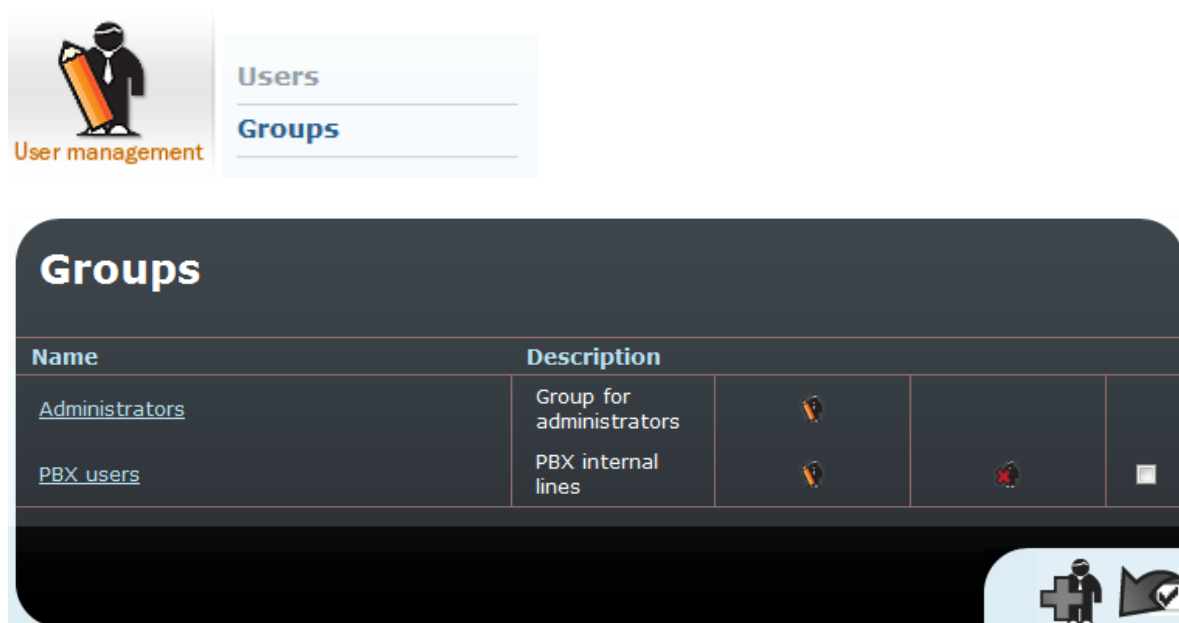
- **User name** – user account name. It must be unique, contain alphanumeric characters only and respect the Upper/Lower Case.
- **New password** – user login password. Dots are displayed instead of characters for safety reasons.
- **Confirm new password** – Since dots are displayed instead of characters, re-enter the password to avoid typing errors.
- **Group** – right group. You can create different right groups for different user groups. To define the range of user rights effectively, simply choose a certain right group. In addition to this, you can tailor the safety rules to a specific user. Refer to the descriptions of the *Rights* and *Rights denied* fields below.
- **Language** – web interface language to be used after the user login.
- **Default application** – the default group tag that is active in the instant of login.
- **Rights** – access rights beyond the selected right group.
- **Rights denied** – rights to be barred to a user although the user should have them because of the selected rights group.

- **Line number** – define the line number under which the IP telephone may be registered to the internal SIP proxy server.
- **Description** – any text that describes the meaning of an account.
- **VoiceMail enabled** – here enable/disable VoiceMail.
- **PIN** - 1111 (default value that can be changed by the user in the voice menu any time later) – used for entering the VoiceMail box from a line other than the user's line.
- **SMS notification number** – GSM number for sending SMS messages notifying that a voice message has been received if the PBX is equipped with a GSM module with a valid SIM card.




4.4.3. Right Groups

The right groups are created for simplification as they eliminate the need to specify the user access rights whenever you create a user account. Upon the system login, the resulting set of assigned and denied rights is composed of the assigned right group settings and, if necessary, corrections made during the user account creation.

To define the right groups, select the *Groups* menu in the *User management* tag. After selection, the list of created groups will be displayed.




The screenshot shows the 'User management' interface. On the left, there is a sidebar with a 'Users' menu and a 'Groups' menu. The 'Groups' menu is selected. The main area displays a table titled 'Groups' with the following data:


Name	Description			
Administrators	Group for administrators			
PBX_users	PBX internal lines			<input type="checkbox"/>

At the bottom right of the table, there is an 'Add user' icon (a person with a plus sign) and a 'Remove selected item' icon (a trash can).

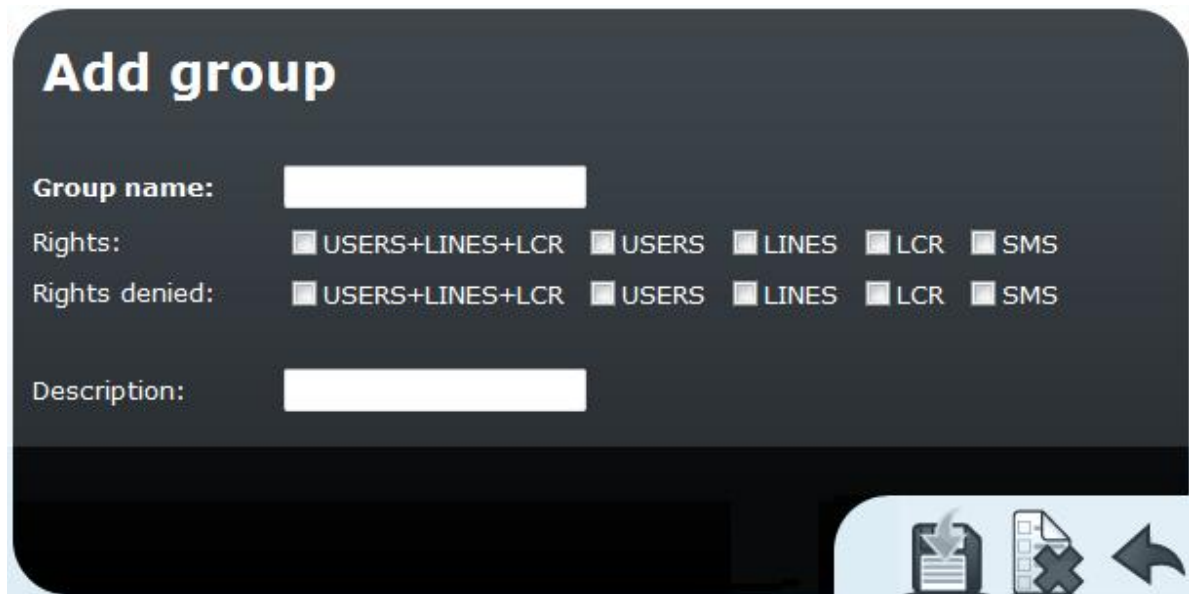
The following items are displayed in the table columns:

- **Name** – right group name.
- **Description** – any text describing the meaning of a group.

Click on the  symbol to move to the group detail for making changes.

The  symbol is used for clearing a right group. To clear a group, you can also tick off the field in the last column of the table and click on *Remove selected item*.

There is an *Add user* icon under the table. Click on it to display the group defining form.



Add group

Group name:

Rights: ☐ USERS+LINES+LCR ☐ USERS ☐ LINES ☐ LCR ☐ SMS

Rights denied: ☐ USERS+LINES+LCR ☐ USERS ☐ LINES ☐ LCR ☐ SMS

Description:

The form contains the following fields:

- **Group name** – unique identification.
- **Rights** – system parts to be made accessible.
- **Rights denied** – system parts to be made inaccessible.
- **Description** – any text describing the meaning of a group.

The web interface is divided into parts where access rights may be assigned or denied. One pair of checkboxes corresponds to each group in the above described right group and user account setting forms. They are as follows:




- **ALL** – whole system.
- **USERS** – user accounts and groups.
- **LINES** – communication lines.
- **LCR** – Least Cost Router configuration.



4.4.4. Ring groups

The **Ring groups** menu helps you create VoIP user groups for incoming call routing.

Use the **User management** tag in the **Ring groups** menu to define ring groups. Open the menu to display the list of available groups.





Ring groups					
Ring group	Extensions	Description			
220	20, 21				

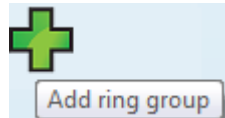



The table columns include:

- **Ring group** – define the group number to enable group ringing.
- **Extensions** – select VoIP users.
- **Description** – type a text to describe the ringing group.

Click on the  (pencil) symbol to open the group details for changes.

To delete a ring group click on the cross , or select the field in the last table column and click on **Remove selected**.



Find the [Add ring group](#) link below the table. Click on it to display a ring group adding form.

Add ring group

Ring group:

Extensions:

Group member

20 ☐

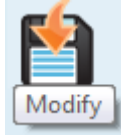
21 ☐






The form includes the following items:

- **Ring group** – define the group number to enable group ringing.
- **Extensions** – a list of all VoIP users that are available for ring group assignment. Select the VoIP users to be assigned to the given group.



Click on  to save the ring group added.

4.4.5. Emergency Change of Administrator Password

If you have forgotten the *Admin* account password, do not despair. Just connect yourself to the module by means of a serial cable and select item 2, *Set Admin password*, in the main terminal console menu. Then, enter a new password and push .

4.4.6. SIP / H.323 Switching

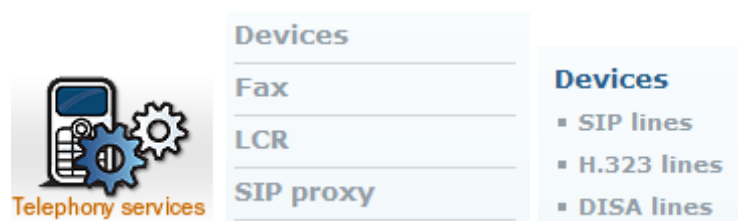
If you are not licensed to use the SIP and H.323 protocols simultaneously, you have to choose one. The default module protocol is H.323.

To toggle the protocols, use the *Switch to SIP/Switch to H.323* option in the following menu. Doing this you also initiate the module restart.

4.5. Communication Line Setting

To define line parameters, use the *Devices* menu in the *Telephony services* tag. You can add and modify the VoIP lines (SIP, H.323 and DISA). To set a line type, select the appropriate *Devices* menu item.

Be sure to establish one SIP or H.323 line at least to make your 2N® OMEGA Lite Ethernet module work properly.



4.5.1. SIP

A clearly arranged table of available SIP lines is displayed on the initial SIP line setting screen:

Devices


- SIP lines
- H.323 lines
- DISA lines


SIP lines

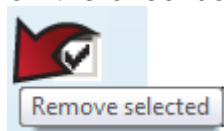
Line ID	SIP server	Phone number	Description			
[12]	192.168.1.215	1111	Moje SIP linka			


The following items are displayed in the table columns:

- **Line ID** – internal line identification used by the module firmware. Click on the number to display the line detail.
- **SIP server** – IP address of the SIP proxy to which the line is being registered.
- **Phone number** – line calling number.
- **Description** – any text describing the meaning of a line.

Click on the  symbol to move to the line setting detail for making changes.

The  symbol is used for clearing a line. To clear a line, you can also tick off the checkbox in the last column of the table and click on *Remove selected* item



There is an *Add SIP line* icon below the table . Click on it to display the line defining form.




4.5.2. SIP Line Parameters

lick on the *Add* icon below the form to add a new line. To modify an existing line, click on *Change*, thus confirming the entered data and requesting the execution of the selected operation.

There are also *Default* and *Back* icons below the form. The first one restores the default value of all fields and the second one returns you to the previous screen.

Modify SIP line ID "12"

SIP server address:	<input type="text" value="192.168.1.215"/>	
SIP domain:	<input type="text"/>	
SIP name:	<input type="text" value="1111"/>	
Display name:	<input type="text" value="PbX"/>	
Listen port:	<input type="text" value="5061"/>	If exists SIP proxy registration, listen port cannot be 5060
User name:	<input type="text" value="siphone"/>	
Password:	<input type="password" value="••••"/>	
Codecs:	<input type="text" value="G.711 A Law 64000 bps"/> <input type="text" value="G.711 u Law 64000 bps"/> <input type="text" value="G.729 8000 bps"/> <input type="text" value="G.723 6300 bps"/> <input type="text"/>	<input type="button" value="▲ Shift up"/> <input type="button" value="▼ Shift down"/>
Add Phone context to REGISTER request:	<input type="text"/>	
Register expires (seconds):	<input type="text" value="0"/>	
Register with proxy:	<input type="checkbox"/>	
Enable CLIP:	<input checked="" type="checkbox"/>	
Allow only one call:	<input type="checkbox"/>	
Get callee from:	<input type="text" value="Request URI"/>	
Enable NAT:	<input type="checkbox"/>	
NAT port begin:	<input type="text" value="0"/>	
NAT port range:	<input type="text" value="0"/>	
NAT IP address:	<input type="text"/>	
No route code:	<input type="text" value="0"/>	
Description:	<input type="text" value="Moje SIP linka"/>	

The form contains the following fields:

- **SIP server address** – IP address or fully qualified network name of the SIP proxy to which the line is to be registered.
- **SIP name** – line calling number.
- **Displayed name** – text to be displayed to the called person.
- **Listen port** – UDP port on which the line receives calls.

- **User name** – line identification for the SIP proxy registration.
- **Password** – authentication password for the SIP proxy registration. Dots are displayed instead of characters for security reasons.
- **Codecs** – list of codecs to be provided by the gateway during voice channel negotiation.
- **Description** – any text describing the meaning of a line.



4.5.3. H.323



A clearly arranged table of available H.323 lines is displayed on the initial H.323 configuration screen:

Devices

- SIP lines
- H.323 lines**
- DISA lines


H.323 lines


Line ID	Call method	Display name	Numbers	Listen port	Description			
[13]	Direct	PbX	1234	1720	Moje H.323 linka			

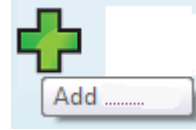



The following items are displayed in the table columns:

- **Line ID** – internal line identification used by the module firmware. Click on the number to display the line detail.
- **Call method** – call initiation method.
- **Displayed name** – text to be displayed to the called person.
- **Numbers** – line calling numbers.
- **Listen port** – port on which the line receives calls.
- **Description** – text entered by the Administrator used for describing multiple lines.

Click on the  symbol to move to the line setting detail for making changes.

The  symbol is used for clearing a line. To clear a line, you can also tick off the checkbox in the last column of the table and click on *Remove selected item*.



There is an *Add H.323 line* icon below the table . Click on it to display the line defining form.

4.5.4. H.323 Line Parameters

To add a new line, use the *Add* icon below the form. To modify an existing line, click on *Change*, thus confirming the entered data and requesting the execution of the selected operation.

There are also *Default* and *Back* icons below the form. The first one restores the default value of all fields and the second one returns you to the previous screen

Add H.323 line

Call method: Gateway ▾

Gatekeeper discovery method: Discover Gatekeeper automatically (use multicast) ▾

Gatekeeper address:

Gatekeeper name:

Gateway address:

Gateway prefix(es):

Endpoint type: Gateway ▾

Display name:

Numbers:

Listen port:

Disable Fast start: ☐

Disable H.245 tunneling: ☐

Disable Early media start: ☐

DTMF type: Signal ▾

Transfer method: Call forwarding ▾

Codecs:




G.711 A Law 64000 bps
G.711 u Law 64000 bps
G.729 8000 bps
G.723 6300 bps

Shift up
Shift down

Rejected call code: Local endpoint application cleared call ▾

No route code: Local endpoint application cleared call ▾

Description:

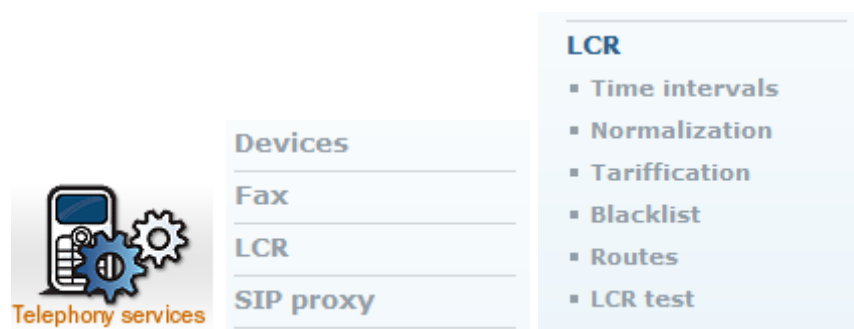
The form contains the following fields:

- **Call method** – call initiation method. There are three options: 1) *Direct* – enter the fully qualified network name of the called person, 2) *Gatekeeper* – a selected Gatekeeper finds the relevant path and establishes the connection; 3) *Gateway* – calls are made via a selected module.

- **Gatekeeper finding method** – determines how to find an available Gatekeeper. Either enter a static address or search the network for the address.
- **Gatekeeper IP address** – here enter the Gatekeeper IP address if the Gatekeeper address is not network-searched.
- **Gatekeeper name** – here enter the Gatekeeper name for line login. This parameter is optional.
- **Gateway address** – here enter the gateway address if you have selected the **Gateway** calling method.
- **Gateway prefix(es)** – prefixes for the gateway login to the Gatekeeper.
- **Endpoint type** – the gateway can login to the Gatekeeper either as a **Gateway** with a prefix (all calls starting with the prefix are routed through the module) or as a **Terminal** (virtual telephone).
- **Displayed name** – text to be displayed to the called line.
- **Numbers** – numbers of the line working in the Terminal mode.
- **Listen port** – TCP port on which the line receives H.323 calls.
- **Disable Fast start** – select this item to disable the **Fast start** method. This method provides a faster speech channel start where the H.323 protocol is used.
- **Disable Early media start** – select this item to disable the **Early media start** method. This method establishes the speech channels before the call has been set up completely by signalling. It is useful for transmission of GSM progress tones.
- **DTMF type** – choose the DTMF tones to be supported; either numbers only (select **Signal**), or alphanumeric characters (select **Alphanum**).
- **Transfer method** – select either **Call forwarding** or **H.452.2 call transfer**.
- **Codecs** – a list of codecs to be provided by the gateway for voice channel negotiations.
- **Description** – any text that describes the meaning of a line.

4.6. Least Cost Router

This section deals with the purpose and operation of the gateway cost saving machine - the Least Cost Router (LCR). Let us explain how it works and can be configured.



Here is a survey of what you will find in this section:

- Least cost routing process;
- Routing rules – destinations, routes, lines and time intervals;
- Number normalising;
- List of barred numbers (Blacklist);
- LCR configuration testing.

The purpose of the Least Cost Router is to find the optimum output line for the called number. The LCR process has several stages:

Input normalising - the calling and called line numbers are transformed into a normalised format before entering the LCR.

Destination searching - a destination means the target party to the call. The destination includes a group and type of line searching in the group. The group is searched according to the prefix of the normalised called number.

Group searching - a group means a logical set of lines. The group definition includes a route time limit within a week.

Line searching - a route can consist of one or more lines. The final line is determined according to the selected line selection algorithm. Three ways of searching are available: 1) *First free line* – finds the first free line, 2) *Cycle* – selects the free line with the earliest time of the last call, and 3) *Free minutes* – uses the free line with the highest number of free minutes. Correct call rates must be selected for the lines for the method to work effectively.

Output normalising - the calling and called line numbers are transformed into a normalised format before the call is transferred to the output line.

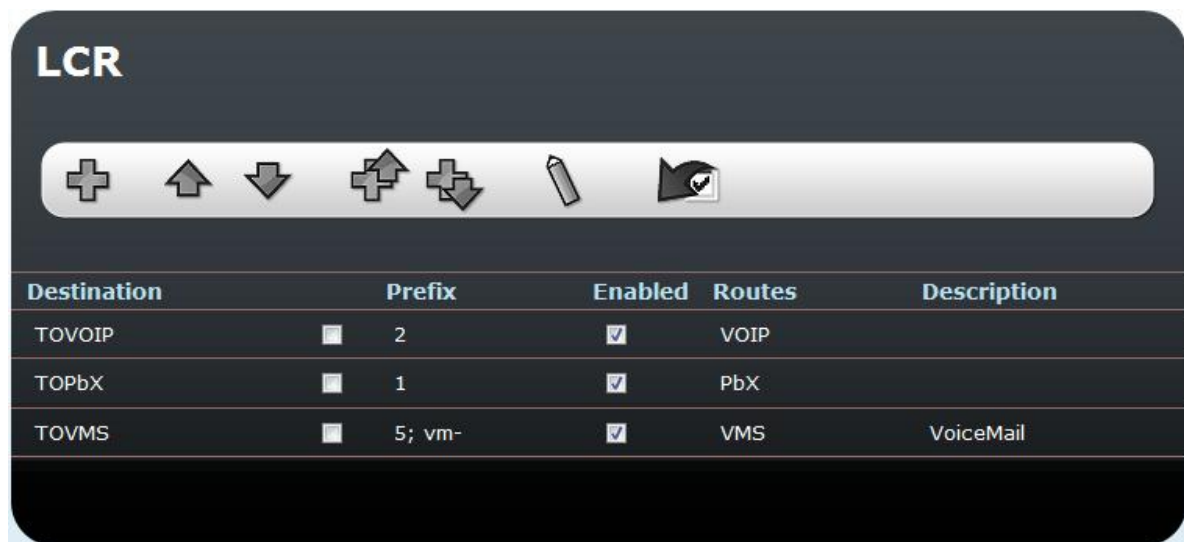
Check for barred numbers - the *Barred numbers* table is searched for match after normalising and before transfer to the successfully found output line, and, if a match is found, the call is rejected.

By default, the LCR connects all VoIP calls to the Ethernet module without modifying the called number.

4.6.1. Routing Rules

Routing rules are the core of the LCR system. They consist of a relatively high number of parameters, which make the routing process highly flexible. A guide is available to make your configuration steps as convenient as possible.

Start with clicking on the *Least Cost Router (LCR)* item in the *Telephony services* group tag menu. A table gets displayed with predefined routing rules and buttons that help you change the sequence, add new items and modify and/or delete the existing rules.



Destination		Prefix	Enabled	Routes	Description
TOVOIP	<input type="checkbox"/>	2	<input checked="" type="checkbox"/>	VOIP	
TOPbX	<input type="checkbox"/>	1	<input checked="" type="checkbox"/>	PbX	
TOVMS	<input type="checkbox"/>	5; vm-	<input checked="" type="checkbox"/>	VMS	VoiceMail

All the above-mentioned buttons except for *Add* are inactive in the default status (greyish). They will not become active until one routing rule at least is selected by ticking off the checkbox in the second column of the respective row. Push the button activated in this way to execute the operation stated above the selected routing rules.

4.6.2. Add Rule

Click on *Add* or *Add after* or *Insert before* to run the LCR adding guide. The guide will be displayed in a new window.



Add LCR table item

Enabled: ☒

Destination name:


Prefix 1 [Add next...](#)

Description:

Route name: [Add route](#)

Line selection:

Description



The first step is to define the destination. Basically, the destination is a set of prefixes. Looking for the appropriate routing rule, the LCR searches the allowed routing rules from top to bottom for a match of the normalised called number prefix and the destination number prefix. The first matching prefix stops the searching process and the call is routed according to the respective destination rule.

The form contains the following fields:

- **Enabled** – A routing rule can be defined yet may keep unused in the routing process. Enabled rules are used only.
- **Destination name** – The destination must be named briefly and clearly, e.g. according to the mobile network provider or any other distinctive feature. The name may contain alphanumerical characters only.
- **Prefix n** – is the destination prefix where n is a serial number. The prefix is a beginning of the target telephone number. If you leave the first prefix blank and do not enter any other, then the particular destination includes all called numbers. To add new prefixes, push the *Add others* key.


- **Description** – any text describing the meaning of a destination.

Push the *Default* button to restore the values available in the instant of form opening. Click on the *Close* key to close the guide window. Push *Next* to proceed to the second part of creation. The *Previous* and *Add* keys are inactive in the first step as they cannot initiate a meaningful operation.

In the second step select the group whose lines are to be used for this rule. To do this, you can either use a list of earlier-defined groups from the pop-up menu or create a new group. Moreover, define how to select the lines included in the group.

4.6.3. Modify Rule

Click on the destination name in the survey of routing rules to display an LCR setting guide, which is similar to that introduced above. The only difference is that the *Add* key is replaced with the *Modify* key.

LCR					
					
Destination		Prefix	Enabled	Routes	Description
TOPBX	<input type="checkbox"/>	1	<input checked="" type="checkbox"/>	PBX	
TOVOIP	<input checked="" type="checkbox"/>	2	<input checked="" type="checkbox"/>	VOIP	
TOVMS	<input type="checkbox"/>	5; vm-	<input checked="" type="checkbox"/>	VMS	VoiceMail



Modify LCR table item

Enabled: ☒

Destination name: TOVOIP

Prefix 1




Prefix 2 [Add next...](#)

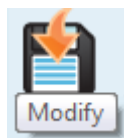
Description:

Route name: [Add route](#)

Line selection:

Description



All the LCR changes influence the gateway behaviour the moment they are executed. Therefore, it is unnecessary to restart the whole system.

4.6.4. Remove Rule

Tick off the checkbox in the second column of the respective row and push the *Remove* button to delete a routing rule.



Destination		Prefix	Enabled	Routes	Description
TOPBX	<input type="checkbox"/>	1	<input checked="" type="checkbox"/>	PBX	
TOVOIP	<input checked="" type="checkbox"/>	2	<input checked="" type="checkbox"/>	VOIP	
TOVMS	<input type="checkbox"/>	5; vm-	<input checked="" type="checkbox"/>	VMS	VoiceMail



4.6.5. Change Sequence


The rule sequence changing procedure is similar to that of the rule removal. Check off the respective row and click on the *Shift up* or *Shift down* key.

4.6.6. Disable/Enable Rule

We already know that only an enabled routing rule may affect the routing process. You can change the *Enabled* status while setting and/or changing a rule. Or, just by clicking on the checkbox in the third column of the respective row in the survey of routing rules too.

4.7. LCR Routes

LCR routes are part of the LCR rules as we mentioned in the section above.

Routes						
Route name	Path lines	Time intervals	Description			
PBX		weekdays weekend workdays				
VMS	DISA - voicemail	weekdays weekend workdays				
VOIP	SIP - Moje SIP linka	weekdays weekend workdays				

4.7.1. List of Routes

Click on the *Routes* item in the *Least Cost Router* submenu to display a list of routes.



4.7.2. Add Route



Click on the *Add* icon to add a route. A form gets displayed in the browser for you to complete the following data:

Add route

Route name:

Lines of route:

SIP - Moje SIP linka
PBX module
DISA - voicemail

▲ Shift up

▼ Shift down

[Add SIP line](#)
[Add H.323 line](#)
[Add SIM card](#)




Time intervals:

weekdays
weekend
workdays

[Add day_group](#)
[Add time_range](#)


CLIP/CLIR: Default ▼ [CLIP routing](#)

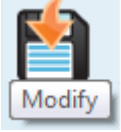

Description:


- **Route name** – The route name may contain any alphanumeric characters. The LCR rules refer to this name as mentioned in the section above.
- **Route lines** – a list of available lines that form a route. Click on the respective item with the left-hand mouse button to select/remove a line. You can select more list items at the same time by pushing the Ctrl key along with the mouse click. The selected items are illuminated blue. Push the *Add SIP line* or *Add H.323 line* key to create new lines.
- **Time intervals** – define the routing rule time validity. There are three time groups by default: *weekdays* – whole days, Monday till Friday, *workdays* – from 7 a.m. to 5 p.m., Monday till Friday, *weekend* – Saturdays and Sundays. Click on the *Add day_group* or *Add time_range* key to create new time intervals.
- **Description** – any text that describes the meaning of a route.

4.7.3. Modify Route

Click on the  symbol of the respective route to display a route setup modifying guide, which is similar to that introduced above. The only difference is

that the *Add* key  is replaced with the *Modify* key .

4.7.4. Remove Route

Click on the  symbol to delete a route. Or, you can also tick off the checkbox in the last table column and click on *Remove selected items*



Remove selected

to delete all selected routes at the same time.

4.8. LCR Number Normalising


Speaking of the LCR operations, we also mentioned number modifications before and after the routing process – the so-called normalising. Normalising means conditioned transformations of the called and/or calling numbers into a unified format, which facilitates definition of the routing rules.

The normalising rule is determined by the following three parameters:

- 1) **Normalising type** – defines at which process stage (input/output) and for which numbers (called/calling) normalising should be used.
- 2) **Condition** – transformation is only applied to numbers that meet a condition. The condition is specified by a prefix – a string that the number starts with. The prefix is separated automatically and the remaining part of the number is transformed only.
- 3) **Transformation rule** – enables to modify a number by removing a certain number of characters from the number beginning, or adding a new prefix.

To set the normalising rules, use the *Normalising* item in the *Least Cost Router (LCR)* menu in the *Telephony services* group tag. The initial screen is a list of created normalising rules.

Click on the  symbol to enter the form and make changes.

To delete a normalising rule, click on the  symbol. To delete all rules, select them by checking the last table column and click on *Remove selected items* below the table.

There is an *Add LCR normalising* icon below the table. Click on it to display the rule defining form.



Add LCR normalization

Line:

Prefix:




Remove count:

Add number:

Type:

Called outgoing ▼

Description:

The meanings of the form fields are identical with the following columns of the normalising table:

- **Prefix** – a prefix that a number must start with to meet the transformation rule.
- **Remove count** – a count of characters to be removed from the number beginning behind the prefix.
- **Add number** – a number to be added before the number part after removal of the prefix and a defined count of digits.
- **Type** – defines the stage at which the normalising rule should be applied. There are four options: *Incoming calling* – on the calling number input, *Incoming called* – on the called number input, *Outgoing calling* – on the calling number output, *Outgoing called* – on the called number output.
- **Description** – any text that describes the meaning of a normalising rule.


4.9. LCR Barred Numbers


The *Barred numbers* table (Blacklist) is searched for match after output normalising and before call forwarding to the successfully found output line. If the normalised number starts with a string included in the table, the call is rejected. This means that we can enter both whole telephone numbers and prefixes into the list to eliminate, e.g., all international calls or special high-rate calls (erotic lines, etc.).

To define the barred numbers, use the *Blacklist* item in the *Least Cost Router (LCR)* menu in the *Telephony services* group tag. A list of barred numbers gets displayed.

The columns include:

- **Regular expression** – the barred number beginning (or the whole barred number).
- **Description** - any text that describes the meaning of an item.

Click on the  symbol to move to the detail and change the settings. The above-mentioned two fields are the only two items to be completed in the modifying form.

To delete a barred number, use the  symbol. Or, tick off the last table column and click on *Remove selected items*.

There is an *Add barred number* icon below the table. Click on it to display the number defining form.

4.10. LCR Testing

The *LCR test* item in the *Least Cost Router (LCR)* menu in the *Telephony services* group tag is used for testing changes in the LCR settings. Enter the

caller's and called numbers and click on the *LCR test* or *Repeat LCR test* icon located below the form to initiate the LCR process simulation.




The image shows a dark-themed form titled "Input LCR test". It contains three input fields: "From line:" with a dropdown menu showing "PBX module - 21", "Caller:" with a text input field, and "Called number:" with a text input field. At the bottom right of the form, there are two icons: a document with a downward arrow and a document with a red 'X'.

After simulation, the normalised calling and called numbers plus the name of the successfully found output line, or the reason why the line search was unsuccessful are displayed.

4.11. SIP Proxy

The 2N® OMEGA Lite Ethernet module is delivered with an integrated SIP proxy, which is able to work as a private branch exchange for SIP telephones. It is easy to configure; all you have to do is fill in the routing table.

SIP lines						
Line ID	SIP server	Phone number	Description			
[12]	192.168.1.215	1111	Moje SIP linka			


Above is an example of a filled-in routing table, which gets displayed after you open the *SIP proxy* menu in the *Telephony services* group tag.



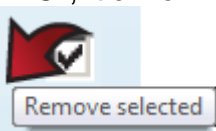
Each row represents one rule. After obtaining the call-establishing request, the SIP proxy searches the table from top to bottom for the relevant routing rule. The relevant rule is based on a comparison of the called destination and the value in the first table column. If the called party's ID starts with the prefix stated in the *If prefix* column of the routing rule, the search is terminated and the call is routed according to the found rule.

The last column in the table represents the default routing rule. It cannot be deleted. it governs all calls for which no explicit routing rule has been found.

The routing rule determines what the SIP proxy should do with a call. A call may be rejected, forwarded to another host and/or port, forwarded to a VoIP line of the module, or routed via the database of available SIP telephones. Before any of the above mentioned operations is executed, the called party's identification may be modified by removing a certain number of characters from the left, or adding a new string to the ID beginning as the case may be (refer to the *Remove* and *Add* columns).

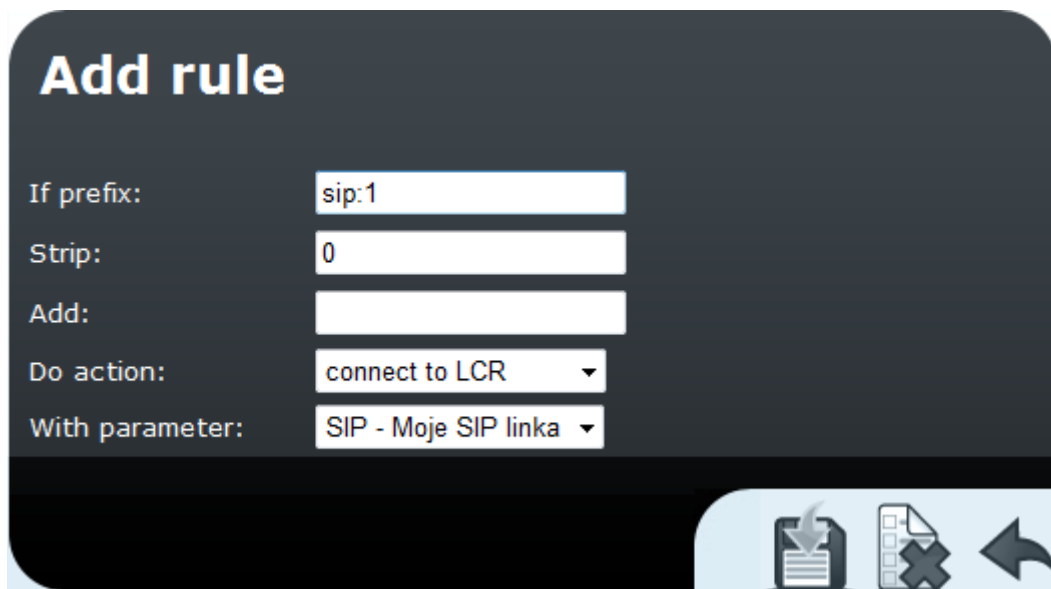
Click on the  symbol to move to the routing rule detail for making changes.

To delete a routing rule, use the  symbol. Or, tick off the last table

column checkbox and click on *Remove selected items* .

There is an *Add rule* icon below the table. Click on it to display the rule defining form.

The figure below shows the SIP proxy routing rule setting form. The form fields correspond to the headers of the above-displayed table.



The form and the table contain the following fields:

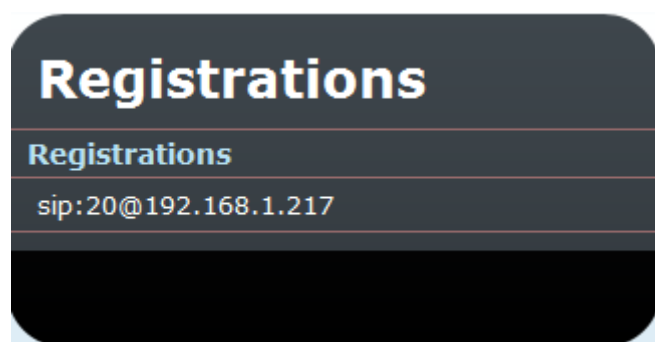
- **If prefix** – if the called subscriber's URI (Uniform Resource Identifier) starts with this string, this rule is used for routing. In the SIP environment, the URI is introduced with the 'sip:' prefix, which has to be included in the value in this field.
- **Remove** – the number of characters following the 'sip:' prefix to be removed from the URI before processing.
- **Add** – the string to be inserted in the URI behind the 'sip:' prefix.

- **Perform action** – there are six possible call actions in the pop-up menu but, basically, three of them are applied only – call rejection, forwarding and interconnection within SIP proxy registrations. Nevertheless, let us mention all of them for completeness: 1) *Reject* – the called line gets the busy tone; 2) *Overwrite host* – forwards the call to the same port of the selected host; 3) *Overwrite port* – forwards the call to the selected port of the same host (the same effect as with option 5); 4) *Overwrite host and port* – forwards the call to any port of the selected host, 5) *Connect to VoIP module* – connects the call to the selected module SIP line, and finally 6) *Search registration* – tries to search the SIP proxy registered users for the required URI and forward the call to the appropriate host.
- **With parameter** – the above actions, except for the first and last ones, require a parameter to be set. This parameter is a new destination for call forwarding and the SIP line name for the *Connect to VoIP module* action.

The call forwarding parameter may be: the host address (2) entered either as an IP address or a fully qualified network name, or the port number (3), or a combination thereof (4) with the host address being separated from the port number by a colon (:).

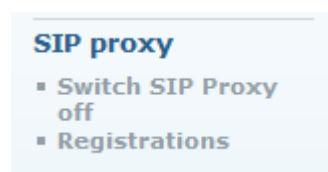
4.11.1. Overview of Registrations

Use the module web interface to see which units have been registered to the SIP proxy. To do this, click on the *Registration* item in the *SIP proxy* menu in the *Telephony services* tag.



4.11.2. Switching ON/OFF

If you are using an external SIP proxy server and do not wish to use the integrated SIP proxy, you can switch the integrated proxy server off.



If ON, the integrated SIP proxy server can be switched off by clicking on the *Switch SIP Proxy off* menu item.

If OFF, the integrated SIP proxy server can be switched on by clicking on the *Switch SIP Proxy on* menu item.

4.12. Mobility Extension

Here the Mobility Extension settings for VoIP phones are described. An authorised Mobility Extension user can make calls as from a VoIP extension. Use the **FLASH = 7*** command during a call for call transfer. Press **9#** to exit the current call, remaining connected to the VoIP and capable of further dialling.

4.12.1. User Settings

Enable the Mobility Extension function and enter the telephone number for the user.



Users

Groups

Mobility Extension

Outgoing ME enabled:

☒

Outgoing ME number:

0601222333

Incoming ME enabled:

☒

Incoming ME number:

601222333

SMS at no answer enabled:

☒

SMS at no answer number:

601222333


Voicemail

Voicemail enabled:

☐

PIN:

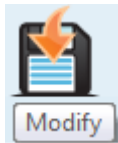
SMS notification number:

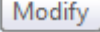


Enter:

- **Outgoing ME enabled** – enable the use of outgoing Mobility Extension, in the case of call to this VoIP user (extension)
- **Outgoing ME number** – the telephone number (including a prefix) to be called in case of an outgoing Mobility Extension call (rings simultaneously with the VoIP extension)

- **Incoming ME enabled** – enable the use of Mobility Extension in incoming ME calls from the user number. The calling user shall be played the “Mobility Extension” voice message and be allowed to dial in the required extension or make an outgoing call from the PBX.
- **Incoming ME number** – the telephone number authorised to use Mobility Extension to call local extensions or external numbers
- **SMS at no answer enabled** – enable sending SMS on missed calls.
- **SMS at no answer number** – enter the telephone number to which SMS on missed calls shall be sent.



Push the  for confirmation.

4.12.2. DISA Line for ME

Create a DISA line for the users that do not meet the Mobility Extension conditions.



Push  to create a DISA line.




Add DISA service line

Type of DISA:

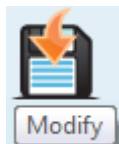
Maximum allowed number of digits:

Operator number:

Description:





  

- **Type of DISA** – select one of the available DISA types. Basic DISA = DISA welcome note is played to the calling user.
- **Maximum allowed count of digits** – enter the count of digits to be dialled within the DISA service.
- **Operator number** – enter the telephone number to be called if the calling user makes no dialling.
- **Description** – an optional brief description.







Push **Modify** for confirmation.

Set DISA line ID

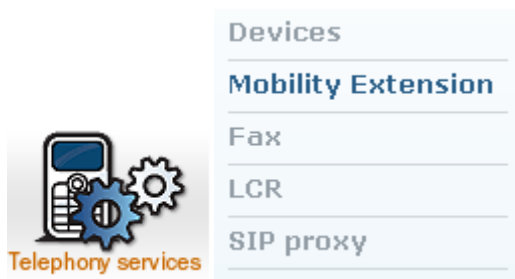
Line ID	Type of DISA	Description		
[22]	Voicemail system	voicemail		
[23]	Basic DISA	Basic DISA		

Data for DISA line successfully set

4.12.3. Mobility Extension




Here define the Mobility Extension conditions.



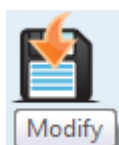
Push  to set the conditions.

Mobility Extension

Outgoing ME enabled:	<input checked="" type="checkbox"/>
Incoming ME enabled:	<input checked="" type="checkbox"/>
Allow sending SMS:	<input checked="" type="checkbox"/>
SMS at no answer text (%n for number):	<input type="text" value="Missed call from %n"/>
Mobility Extension SIP route:	<input type="text" value="192.168.1.1:5061"/>
Mobility Extension DISA line:	<input type="text" value="DISA - 23"/>

- **Outgoing ME enabled** – tick off to enable a ME outgoing call.
- **Incoming ME enabled** – tick off to enable ME in an incoming call.
- **Allow sending SMS** – tick off to enable sending SMS on missed calls (via the first GSM port).
- **SMS at no answer text (%n for number)** – type the SMS text to be sent after a missed call. %n stands for the calling number.
- **ME SIP route** – enter the IP address including the port on the device that “controls” the ME function. As a rule, the own Ethernet module IP address is used.
- **ME DISA line** – select the DISA line to be used for ME.



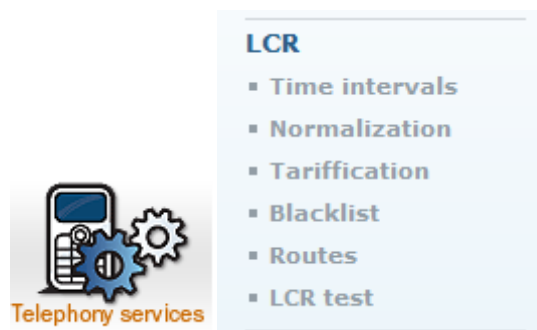
Push  for confirmation.

Mobility Extension

Outgoing ME enabled:	<input checked="" type="checkbox"/>
Incoming ME enabled:	<input checked="" type="checkbox"/>
Allow sending SMS:	<input checked="" type="checkbox"/>
SMS at no answer text (%n for number):	Zmeškané volání od %n
Mobility Extension SIP route:	192.168.1.1:5061
Mobility Extension DISA line:	23

4.12.4. LCR – Least Cost Router


Here define the calling conditions for the Mobility Extension function.



4.12.4.1. Routes

Create routes for Mobility Extension.



Push  to create a line in LCR Routes.

Add route

Route name:

Lines of route:

DISA - Basic DISA

SIP - Moje SIP linka

PBX module - 21

DISA - voicemail

▲ Shift up

▼ Shift down

[Add SIP line](#)
[Add H.323 line](#)
[Add SIM card](#)

Time intervals:

weekdays

weekend

workdays




[Add day group](#)
[Add time range](#)

CLIP/CLIR:

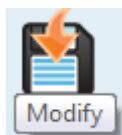
Default ▼

[CLIP routing](#)


Description:

- **Route name** – enter the name of the route to be presented in other LCR sections.
- **Route lines** – select the DISA line to be used for the ME function.
- **Time intervals** – specify when the route may be used.
- **CLIP/CLIR** – select Default.
- **Description** - an optional brief description.



Push  for confirmation.

Routes						
Route name	Path lines	Time intervals	Description			
mobility	DISA - Basic DISA	weekdays weekend workdays				
PBX	PBX module - 21	weekdays weekend workdays				
VMS	DISA - voicemail	weekdays weekend workdays				
VOIP	SIP - Moje SIP linka	weekdays weekend workdays				

4.12.4.2. Normalising

Normalising has to be made for incoming calls to the Ethernet module to adjust the called number to the Mobility Extension conditions (if the calling party number doesn't match to an incoming ME number of any user). In fact, to set correctly the ME function, called number that comes from the basic part (e.g. from GSM) has to be empty (see 4.12.5.1). Here it has to be changed to a number that could be use in LCR rule. This new number will be used ONLY if the caller (calling party) number doesn't match to any incoming ME number.



Telephony services




LCR

- Time intervals
- **Normalization**
- Tariffication
- Blacklist
- Routes
- LCR test

Push  to create a LCR normalising condition.

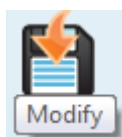
Add LCR normalization

Line: PBX module - 21 ▾
Regular expression: ^\$
Rule: 444
Type: Called incoming ▾
Description:

This format of Add LCR normalization menu is set by parameter “Advanced config of regular expressions” (see 4.4.1).

- **Line** – select the incoming line to which normalising shall apply.
- **Regular expression** – set a normalising condition to match to incoming calls with an empty called number (no called number was passed to the Ethernet module). The correct syntax of the regular expression is ^\$ in this case.
- **Rule** – enter which number shall be “dialled” after the condition is met. In LCR this called number will be routed to DISA.
- **Type** – define the call type for the condition. Select Called incoming.
- **Description** - an optional brief description.



Push  for confirmation.

If the call is recognized as an incoming ME call, the empty called number is automatically replaced with string #mobext#. No normalising rule is needed.

Normalization					
Line	Regular expression	Rule	Type	Description	
<input type="checkbox"/>	^5	vm-5R	Called outgoing		
<input type="checkbox"/> 21	^5	444	Called incoming		

4.12.4.3. LCR Prefix

Here define the Least Cost Router (LCR) conditions for the ME function.



Push  to create a LCR condition.

Add LCR table item

Enabled: ☒

Destination name: mobility

Prefix 1: 4

Prefix 2: #mobext#


Prefix 3: [Add next...](#)

Description:

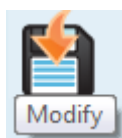
Route name: mobility [Add route](#)

Line selection: Default

Description:



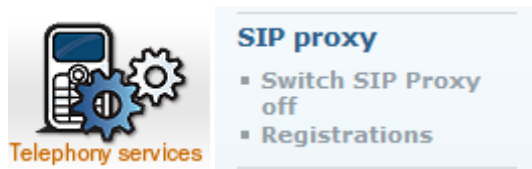
- **Destination name** – name of the destination to be displayed.
- **Prefix 1** – enter the prefix for the calls that do not meet the ME condition i.e. the calling number does not match any of the incoming ME numbers set for the users. It should correspond to the number set in the “Rule” parameter (see 4.12.4.2)
- **Prefix 2** - enter the prefix for the calls that meet the incoming ME condition. The correct syntax is **#mobext#**.
- **Description** - an optional brief description.
- **Route name** – select the earlier created route for the ME function.
- **Line selection** – select the line type (Default).
- **Description** - an optional brief description.



Push  for confirmation.

4.12.4.4. SIP Proxy

Here assign prefixes to the SIP Proxy.

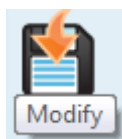


Push  to create a SIP Proxy rule.

A screenshot of a dark-themed 'Add rule' form. It contains several input fields and dropdown menus: 'If prefix:' with 'sip:0', 'Strip:' with '0', an empty 'Add:' field, 'Do action:' with a dropdown set to 'connect to LCR', and 'With parameter:' with a dropdown set to 'SIP - Moje SIP linka'. At the bottom right, there are three icons: a document with a downward arrow, a document with an 'X', and a left-pointing arrow.

- **If prefix** – select which prefix shall affect this rule. Here the rule applies to prefix '0'.
- **Strip** – define how many digits are to be removed. Here none is removed.
- **Add** – define which numbers are to be added.
- **Do action** – define which action is to be performed. Select 'connect to LCR'.
- **With parameter** – define which line is to be used.

Take the same steps to create the condition for prefix '4'.



Push  for confirmation.

SIP proxy routing rules							
If prefix	Strip	Add	Do action	With parameter			
sip:1	0		connect to LCR	SIP - Moje SIP linka			
sip:5	0		connect to LCR	SIP - Moje SIP linka			
sip:4	0		connect to LCR	SIP - Moje SIP linka			
sip:0	0		connect to LCR	SIP - Moje SIP linka			
else	0		lookup registration				

Now the rules for ME application in the Ethernet module configuration tool have been created.

The next step is to configure the PBX using the PBX configuration tool.

4.12.5. PBX Settings

You are advised to use the GSM ports in the TDM (digital) section of the PBX to make the VoIP phone execute the ME function reliably. Use the 2N® OMEGA Lite PBX configuration tool to set the required parameters.

4.12.5.1. Incoming Calls to VoIP from PBX TDM

Define that an “empty” number is dialled to VoIP to enable incoming calls from the TDM section of the PBX. To do this, use the ‘I am calling short dialling’ function in the ringing table.

Set the short dialling as follows:

Short dials/Common tag

Short dials common			
Text	Number		
0	To VoIP	,	
	Call Type	Trunk 6	
			<input type="checkbox"/> Not able for group 1 <input type="checkbox"/> Not able for group 2 <input type="checkbox"/> By-pass authorization <input type="checkbox"/> Not able to delete
0-24 25-49 50-74 75-99 100-124 125-149 150-174 175-199			Free memory: 32754 99%

- **Text** – enter the name of short dialling item to be presented to the user and in the ringing tables.
- **Number** – enter the ‘comma’ character “,”, which represents a pause, to make sure that a pause with no further dialling will be dialled into the trunk specified in the Call type. It will be decided in the VoIP section (Ethernet module) according to the CLIP whether the calling user will be assigned ME. If CLIP shouldn’t match to defined incoming ME numbers, the call will be routed to DISA. In that case, caller would have two options - to dial in the required VoIP phone or wait for the VoIP phone designated as the operator.
- **Call type** – select the bundle (trunk) of VoIP lines (voice channels).

Create a ringing table and add it to the incoming line.

Use the **External lines/Ringing** menu to create a ringing table.

Ringing tabs		
	Action	Par.
1	Call short dial	To VoIP
2	Hang up	---

Influence
Direct: G5m:1

Erase

Assign the port of the internal/virtual line to which the calls are to be accounted to the incoming lines in the **External lines/Transit call identification** menu.

Transit call identification

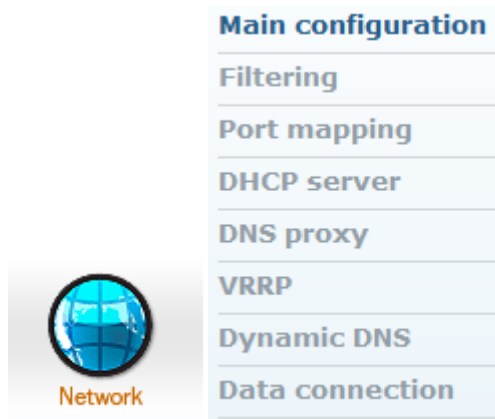
GSM1	11
GSM2	---
GSM3	---
VoIP	---

4.13. PPPoE

Provides to connect Omega Lite to Internet over ADSL modem. Ethernet modul could be connected to ADSL modem or ADSL router set to the bridge mode (without router functions). In that case Ethernet module is set as a router. It could be used just to connect Omega Lite to Internet or could play a role of the LAN gateway. In the next chapters you can found the procedure how to set Ethernet module as a LAN gateway (ADSL router).


4.13.1. Main configuration

The PPPoE setting has to be done in the basic network settings.



It is supposed that the default network setting is used and LAN PCs are set to get IP address from DHCP and DNS servers IP addresses automatically.



Push  to set the conditions.

The default setting is :

Modify network configuration

eth0

Enable DHCP: ☐

IP address:

Network mask:

Gateway:

PPPoE

PPPoE enabled: ☐

Username:

Password:

DNS


Hostname:

DNS server 1:

DNS server 2:

Domain:

Note: Modify network configuration take some time, please wait for result after *Modify* button press.



Set PPPoE items:

PPPoE

PPPoE enabled: ☐

Username:

Password:

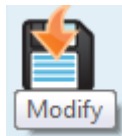
- **PPPoE enabled:** enable PPPoE (Point to Point over Ethernet).
- **Username:** - user name (provided by ADSL provider).
- **Password:** - password (provided by ADSL provider).

PPPoE

PPPoE enabled: ☒

Username:

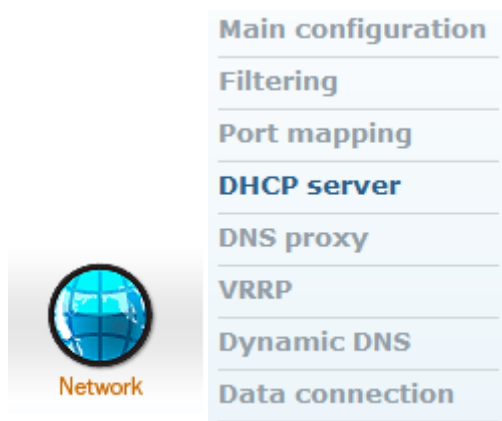
Password:




Push  for confirmation.

4.13.2. DHCP server settings

DHCP server will be enabled in order to provide IP addresses to LAN PCs. The following settings are requested.



Push  to set the conditions.

Modify configuration

DHCP server enabled: ☐

Lease time:

Start address:

End address:


Subnet option:

Router option:

Primary DNS option:

Secondary DNS option:




Domain option:

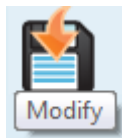


- **DHCP server enabled:** - enables DHCP server i.e. activate settings of the other parameters in this menu.
- **Lease time:** - validity of a leased IP address in seconds.
- **Start address:** - start of an IP address pool.
- **End address:** - end of an IP address pool.
- **Subnet option:** - subnet mask for an internal network (LAN)
- **Router option:** - the default gateway of the subnet (typically the Ethernet module IP address, for VRRP it is the IP address of the virtual router).
- **Primary / Secondary DNS option:** - the IP address of DNS server that will be used in LAN PCs (device). Typically it is the IP address of Ethernet module. Other options are IP address of the local DNS server or IP address of DNS server recommended by IP operator.
- **Domain option:** - domain name of ethernet module. It could be any, but it is requested. Empty field could make problem to PCs running some OSs.

Modify configuration

DHCP server enabled:	<input checked="" type="checkbox"/>
Lease time:	500
Start address:	192.168.1.2
End address:	192.168.1.10
Subnet option:	255.255.255.0
Router option:	192.168.1.1
Primary DNS option:	192.168.1.1
Secondary DNS option:	
Domain option:	OmegaLite



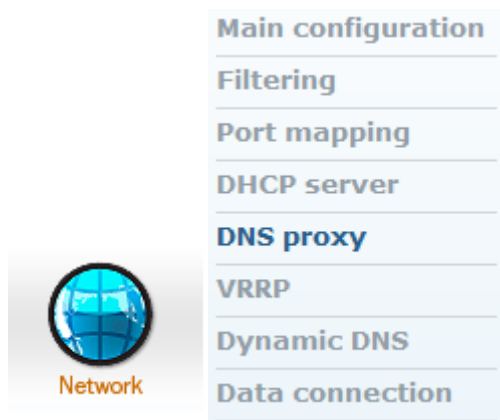


Push  for confirmation.

4.13.3. DNS proxy

Used for caching DNS records in case Ethernet module works as a UMTS/ADSL router. The DNS proxy can receive DNS queries from the local network and forward them to an Internet Domain Name Server. If the DNS Proxy is disabled, the gateway forwards all requests to the servers assigned by the UMTS/ADSL provider.

The Ethernet module can use the DNSs from the *Network settings* or you can forward the DNS queries to specified servers.



Push  to set the conditions.




Modify DNS proxy configuration

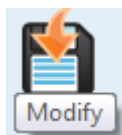
Enable DNS proxy: ☒

Use DNS from network settings: ☒

DNS server:

DNS server 2:





Push  for confirmation.

This was the last step in setting Ethernet module as an ADSL router.

Network config

Enable DHCP:



IP address:

192.168.1.1

Network mask:

255.255.255.0

Gateway:

0.0.0.0

eth0

PPPoE

PPPoE enabled:



Username:

O2

Password:

ppp0

Local IP address:

83.208.22.56

Remote IP address:

88.103.200.41

DNS

Hostname:

voiceblue

DNS server 1:

160.218.161.54

DNS server 2:

194.228.41.65

Domain:

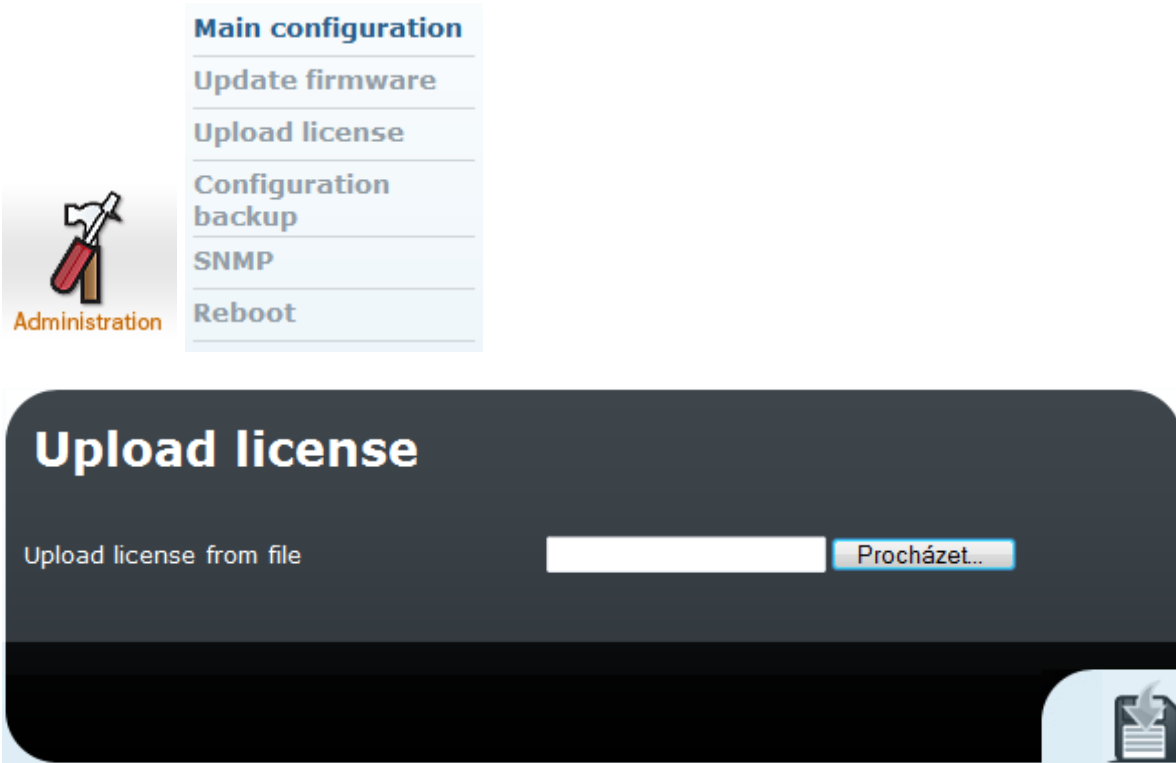


4.14. Licence File

The licence file is used for activating the services provided by the 2N® OMEGA Lite Ethernet module. A brand new module provides 800 hours of operation without the licence file.

Before uploading, save the licence file on your PC hard disk and remember the path to it.

Click on the *Upload licence* item in the *Administration* menu to upload the licence file into the module. This displays the licence file adding dialogue as shown below.



The image shows a web interface. On the left, there is a sidebar with a wrench and screwdriver icon and the word "Administration" below it. To the right of the sidebar is a menu with the following items: "Main configuration", "Update firmware", "Upload license", "Configuration backup", "SNMP", and "Reboot". Below the menu, there is a large dark grey dialog box titled "Upload license". Inside the dialog, there is a text input field with the placeholder "Upload license from file" and a button labeled "Procházet...". In the bottom right corner of the dialog, there is a small icon of a document with a downward arrow.

Push the *Browse* button to select the path to the licence file and the *Add* button to upload the licence file into the module.

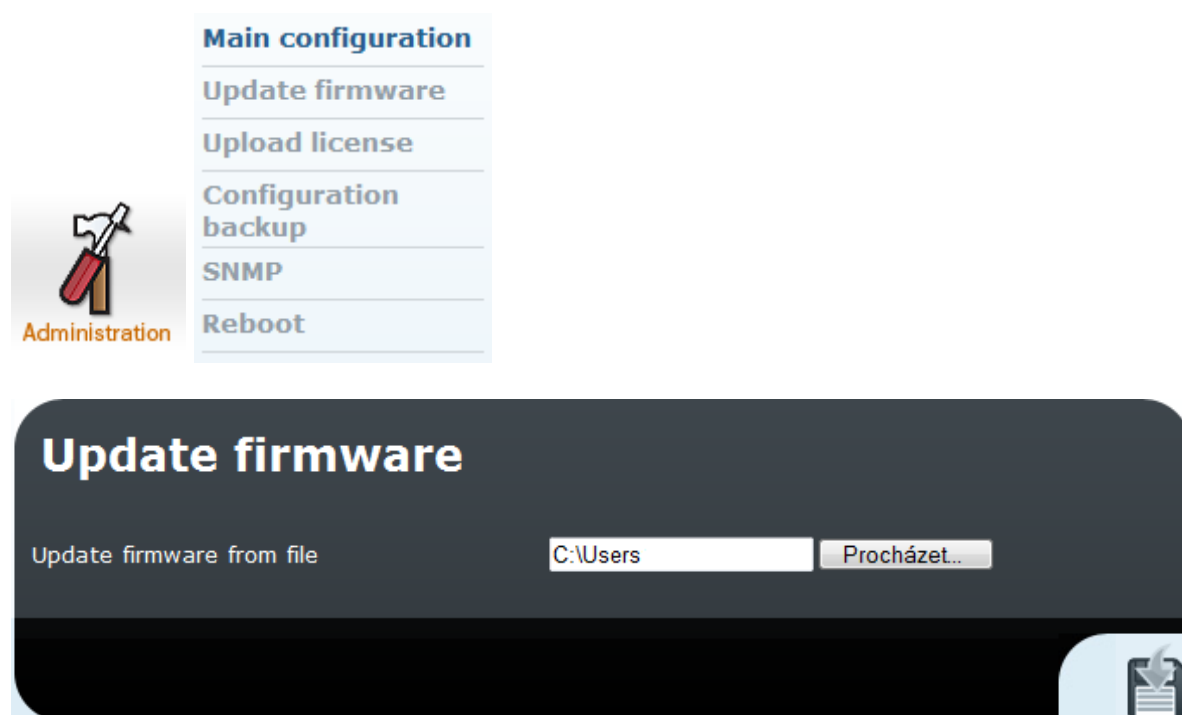
4.15. Firmware Updating

The firmware updating function helps you update the 2N[®] OMEGA Lite Ethernet module firmware. Be sure to use the firmware supplied by the module producer or the producer's technical support department only.

During firmware updating, new firmware versions with innovated functions are downloaded.

The firmware file is named root.tgz.gpg and should have approximately 10 MB. The firmware updating process takes 5-10 minutes if you use your LAN and a delay may occur due to a lower transmission rate if you use the Internet.

Click on the *Update firmware* item to update the firmware. The firmware updating dialogue is displayed as shown below.

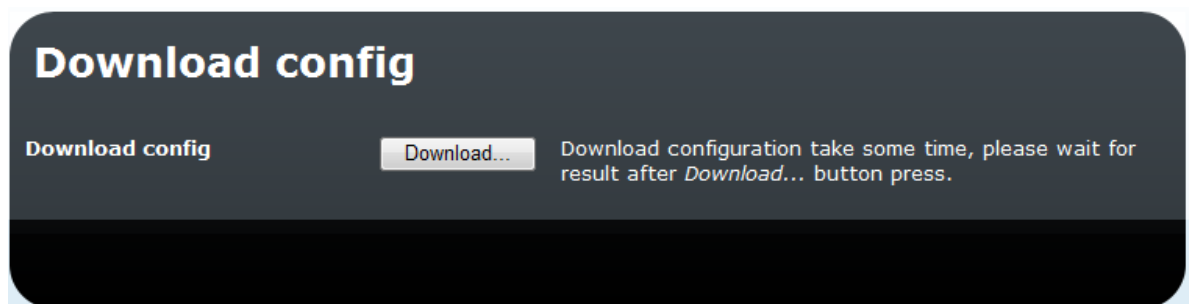
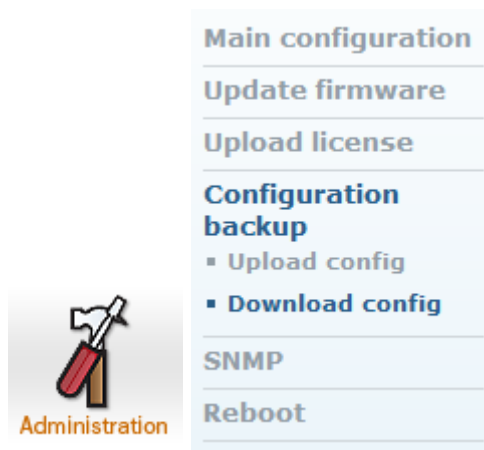


Push the *Browse* button to select the path to the firmware file and the *Update* button to upload the firmware file into the module.

4.16. Configuration Downloading

You can download any configuration from the Ethernet module and save it into a file.

To do this, click on the *Configuration backup/Download config* item in the *Administration* tag. The configuration downloading dialogue will be displayed.

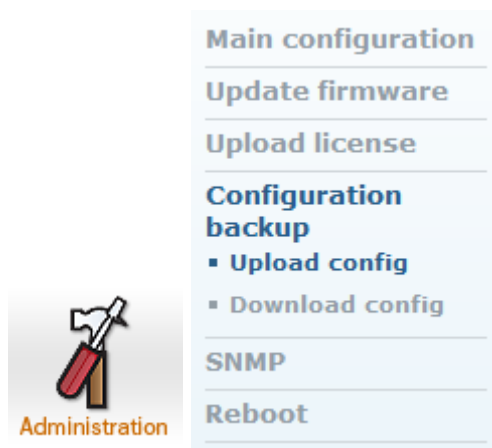


Save the successfully downloaded configuration *backup.tgz* to an unambiguous destination.

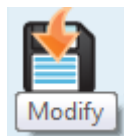
4.17. Configuration Uploading

You can upload the saved configuration into the Ethernet module.

To do this, click on the *Configuration backup/Upload config* item in the *Administration* tag. The configuration uploading dialogue will be displayed.



Find the saved configuration *backup.tgz* and push *Modify* to upload the configuration.



The downloaded configuration also includes the licence effective as of the backup date. Select *Do not extract licence* before uploading to keep the current licence valid.

5. Ethernet Module Parameter Setting by PBX Configuring Tool

The 2N[®] OMEGA Lite PBX has to be equipped with the firmware version 4.02REV11 or later for the Ethernet module to work properly. If not, upgrade your PBX.

5.1. 2N[®] OMEGA PBX Firmware Upgrade

The proper firmware upgrading procedure is as follows:













1. Read and save the PBX settings by the original PBX configuration tool.
2. Read all accounting data (if they are to be further processed).
3. Terminate all open connections.
4. Use the original configuration tool to import the firmware in the Data – Saving software in the PBX menu and reset and boot from flash (you will be prompted automatically by the configuration tool).
5. Reset the factory values.
6. Launch the current configuration tool and create a new configuration (while upgrading versions lower than 4.02), or download the originally stored configuration).
7. Check all the PBX settings.
8. Back-up the configuration on the hard disk.
9. Export the data to the PBX.

5.2. Hardware Configuration

Having performed the PBX firmware upgrade, insert and connect the VoIP module (with the PBX switched off). After the PBX power up download the data and adjust all the module and PBX settings.

After downloading, the presence of the Ethernet module is indicated in the *Global data / Information* in the configuration tool and in the *Global data/Hardware* module.

Information																	
PBX 2N OMEGA Lite																	
Version:	4.02	Revision:	11														
Serial No.:	0609/50006	RA Validity:	198														
Board No.:	06-1411-0049																
SW valid:	Neomezeno	ME valid:	---														
Last license:	---	Last license:	---														
		ME count:	2														
RAM :	V4.02EN.Lite.20070731.1700.Beta.Rev.11.bin																
FLASH :	V4.02EN.Lite.20070731.1700.Beta.Rev.11.bin																
<table border="1"> <tbody> <tr> <td>IL :</td> <td>6</td> </tr> <tr> <td>SYS :</td> <td>2</td> </tr> <tr> <td>AUDIO :</td> <td>0</td> </tr> <tr> <td>CO :</td> <td>0</td> </tr> <tr> <td>ISDN :</td> <td>2</td> </tr> <tr> <td>GSM :</td> <td>2</td> </tr> <tr> <td>VoIP :</td> <td>8</td> </tr> </tbody> </table>				IL :	6	SYS :	2	AUDIO :	0	CO :	0	ISDN :	2	GSM :	2	VoIP :	8
IL :	6																
SYS :	2																
AUDIO :	0																
CO :	0																
ISDN :	2																
GSM :	2																
VoIP :	8																
Last update:		User:															
		SuperVisor:															

PBX HW configuration	
	2N OMEGA Lite
	Basic desk
	Ex Extender 1
	VoIP
	Licenced
	Licenced
	Licenced
	Licenced
	Licenced
	Licenced
	Licenced
	Licenced

5.3. VoIP Line Setting

Use the *External lines/Digital line types/VoIP* in the PBX configuration tool for setting the VoIP line.

The screenshot shows a software window titled "Digital line types" with a blue header bar. Below the header is a tabbed interface with tabs for "Line", "Name", "Options", and "Ringing tabs". The "Options" tab is currently selected. Inside this tab, there are several configuration fields: a text box labeled "VoIP gate" containing "VoIP gate"; two checkboxes, "Without autoriz." and "Private", both of which are unchecked; a numeric spinner for "Min.time for accounting [s]" set to "1"; a text box for "Prefix" which is empty; a numeric spinner for "Priority" set to "0"; a text box for "CLIP" containing "811"; a numeric spinner for "Strip" set to "3"; and a numeric spinner for "Dial to VoIP [s]" set to "2.5". At the bottom of the window, there are two tabs: "So1" and "VoIP", with "VoIP" being the active tab.

Set the following:

- **Name** – line name of up to 14 characters.
- **Without authoris.** – no authorisation check is performed on this line.
- **Private** – the line is connected to a private network where no accounting is performed.
- **Prefix** – prefix of up to four digits, which is automatically called after outgoing line seizure as the first (DID via the superior exchange). The prefix is not written into the accounting line.
- **Minimum accounting time** - time subtracted from the call time for pseudo-accounting.
- **Priority** – an internal line must be of a higher or equal priority to use an external line.
- **CLIP** – enter how the line should be identified for outgoing calls.
- **Remove** – enter the number of digits to be removed from the incoming identification in the inbound direction in order that the remaining digits are the DID to the subscriber or the ringing table.
- **Dial to VoIP [s]** – maximum delay before the next VoIP line dialling. The timeout is reset after every digit received from the internal subscriber and the dialling mode is changed into the calling mode after its expiry. If enabled, the expiry and thus the dialling end are announced by a short beep. Any later dialling is considered a service into the call.

5.4. Outgoing Calling

The VoIP lines have to be bundled (trunked) for outgoing calls. In the factory setting, the VoIP lines are automatically included in trunk 6 and all internal lines may seize the trunk using the *Seizure of external line trunk* (dialling 86).

Include the VoIP lines into the LCR for comfortable calling.

Trunks						
	TRUNK1	TRUNK2	TRUNK3	TRUNK4	TRUNK5	TRUNK6
1	So1	CO1	---	GSM1	So1	VoIP
2	So2	CO2	---	GSM2	So2	---
3	CO1	---	---	---	---	---
4	CO2	---	---	---	---	---
5	GSM1	---	---	---	---	---
6	GSM2	---	---	---	---	---

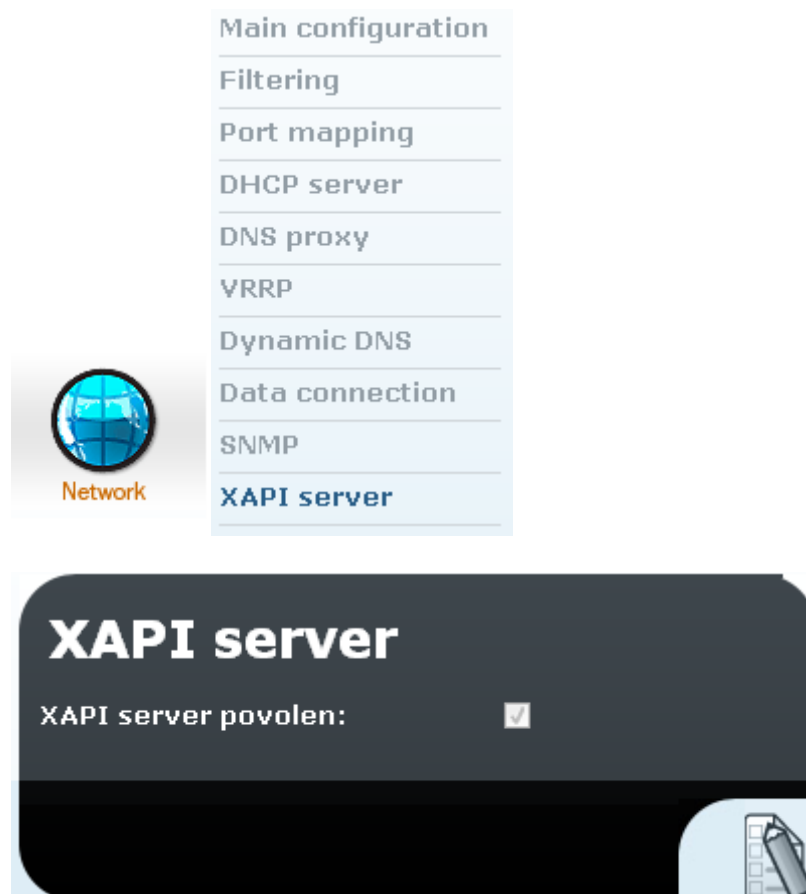
5.5. Incoming Ringing

The behaviour of the VoIP module is the same as that of the ISDN BRI module with the DDI function in the inward direction. Unless specified in the module setting, define the count of digits to be removed from the incoming identification, the remaining digits being considered as the dial-in. Then, the DID is used for the PBX subscriber according to the DID numbering plan:

Or, the DID ringing table is used as the case is with the ISDN dial-in.

6. 2N[®] XAPI Server

The Ethernet module also integrates a 2N[®] XAPI server, which helps the PBX users work with the available applications (PBX Assistant, SMS server, TAPI client, etc.). For details on the applications refer to the installation CD enclosed to the Ethernet module or PBX.



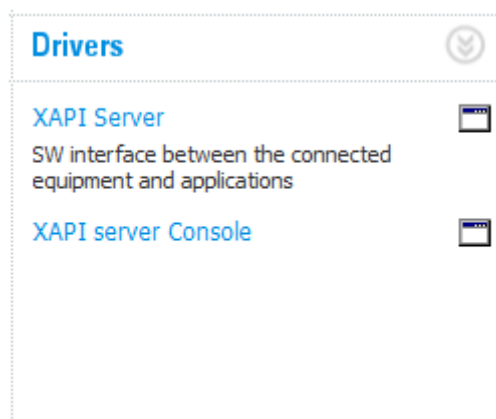
6.1. 2N[®] XAPI Server Console

The XAPI Server Console is designed for 2N[®] XAPI server administration. It helps you create application users (clients), add licences, download accounting and system data, set the input number normalising for the running user applications, and so on.

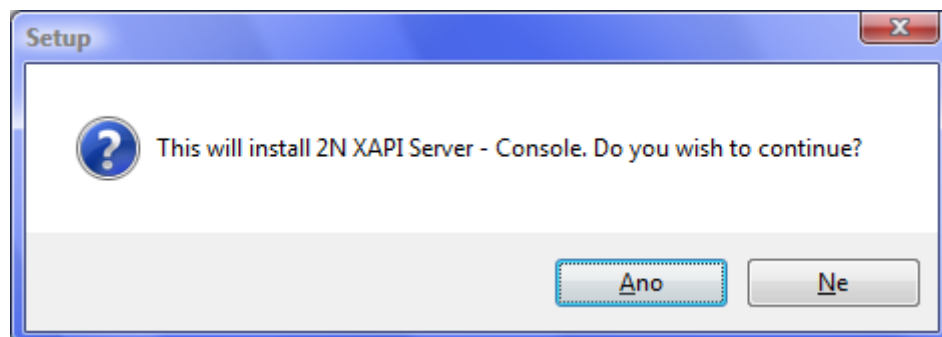


6.1.1. Installation

For installation use the available installation CD and a PC, or your PBX system administrator's server.



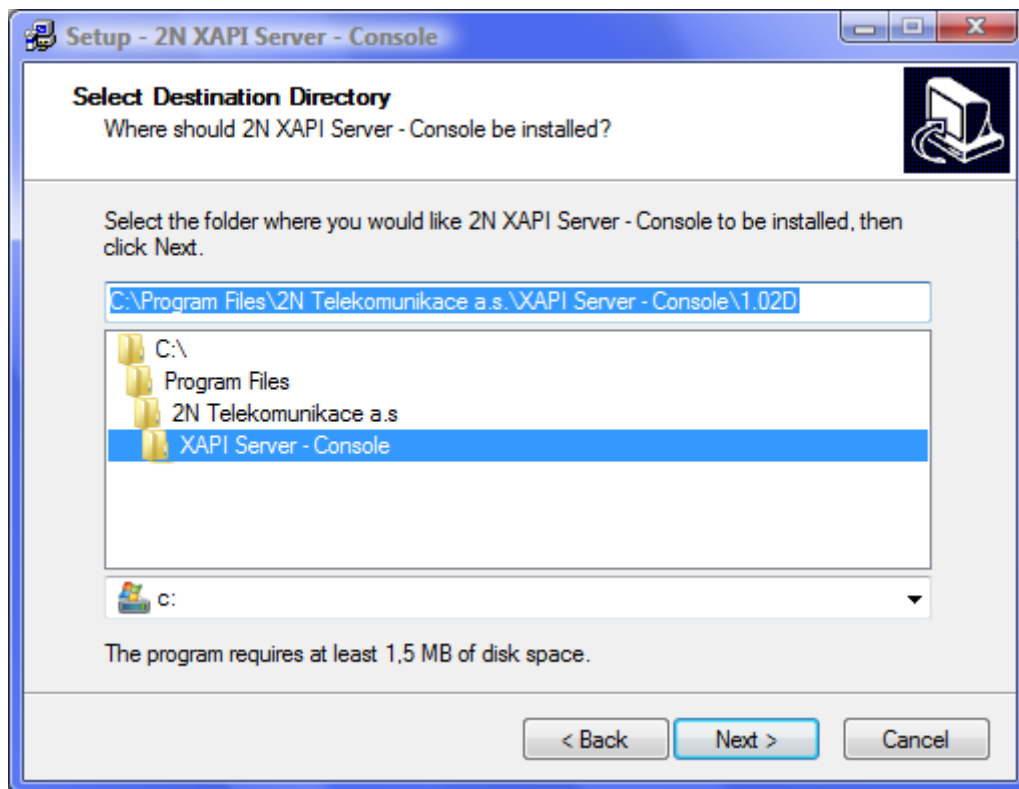
Select the XAPI Server Console and enable installation:



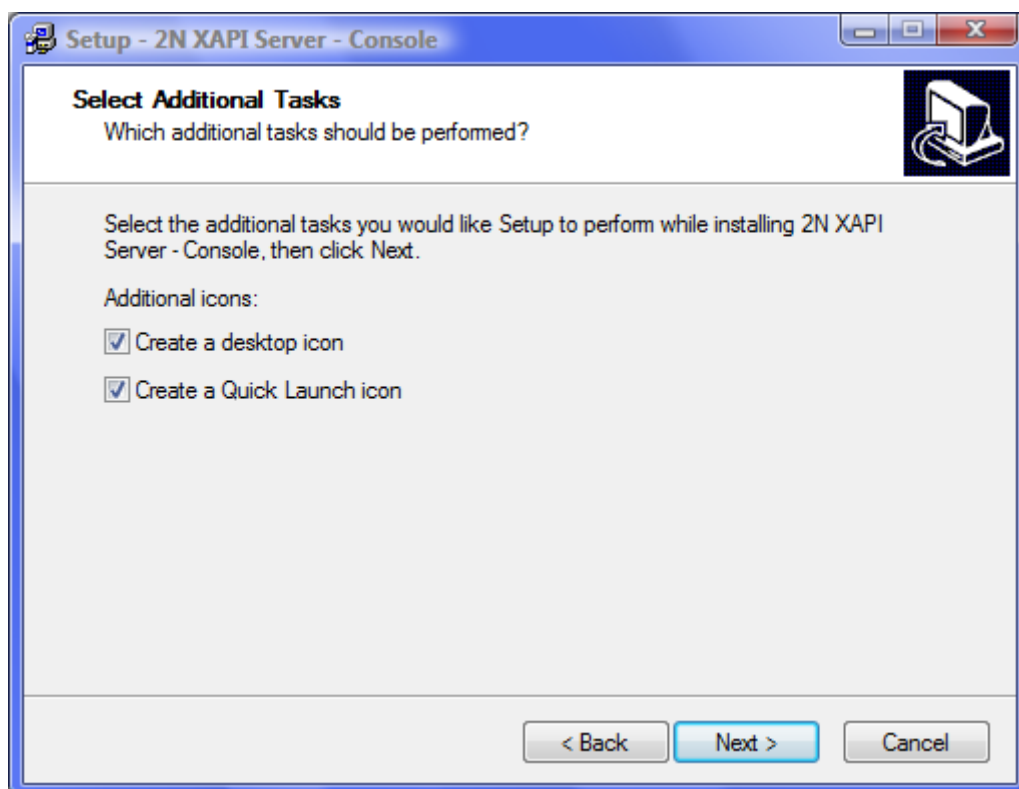
Push Yes:

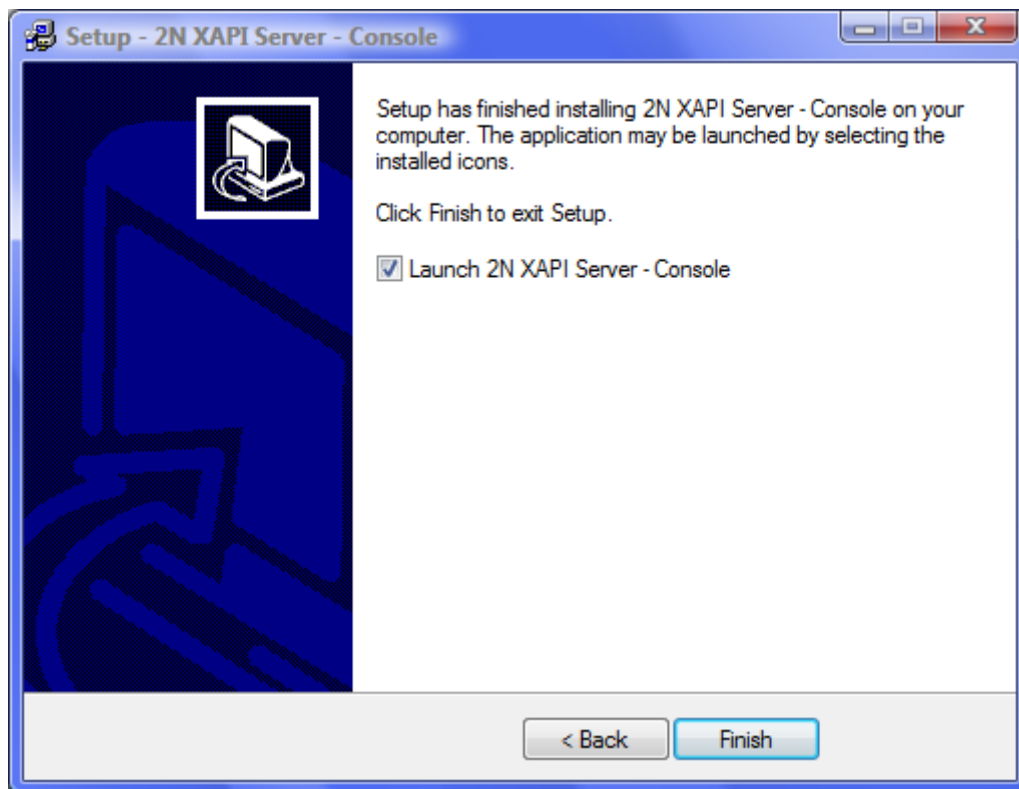


Push Next:



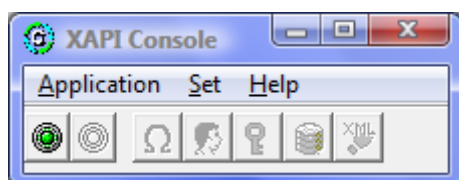
Select the target directory and complete installation.



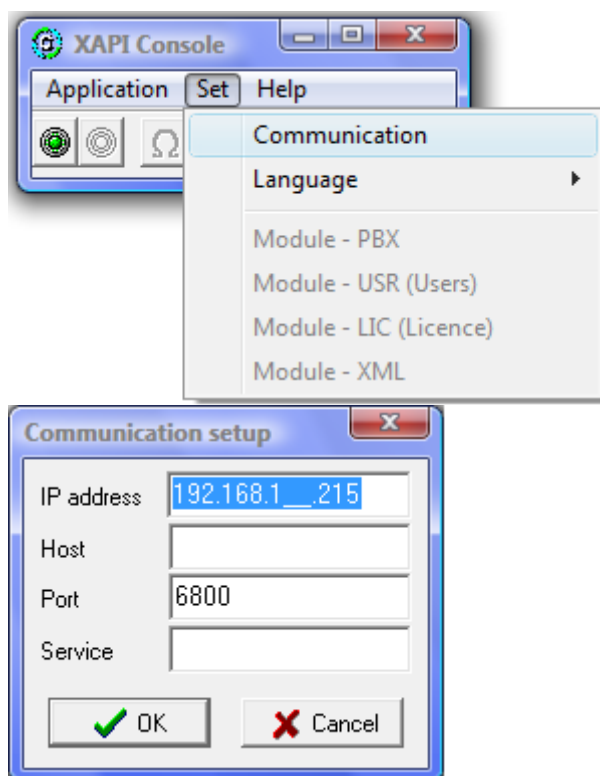


6.1.2. Communication Setting

If you run the 2N[®] XAPI Server Console for the first time, set its communication with the running 2N[®] XAPI server.



To set communication, use the *Set/Communication* menu.

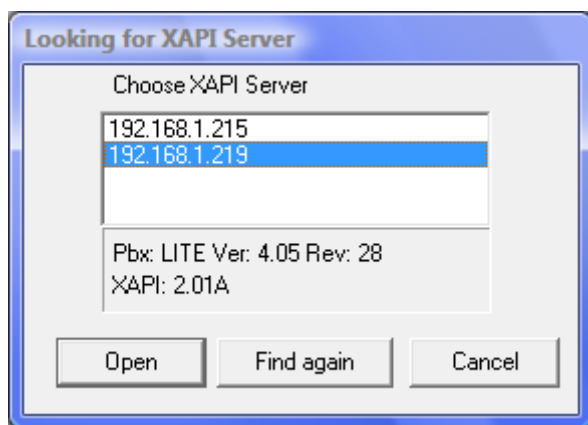


Enter the following:

- **IP address** – IP address of the running 2N[®] XAPI server (Ethernet module in this case).
- **Host** – name of the running 2N XAPI server PC (LocalHost), entered instead of the IP address.
- **Port** – 2N[®] XAPI server communication port. The default value is 6800 and there is no need to change it.
- **Service** – registered service number – to be entered instead of the port.

Push *OK* for confirmation.



If the IP address setting is incorrect, active XAPI servers are searched for automatically within the LAN and you can select the IP address of the required connection directly.

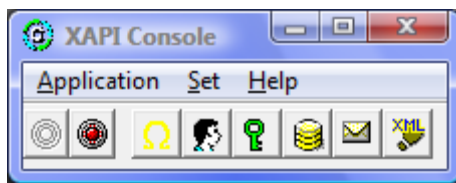


Push *Open* to open connection.
Push *Find again* to start another search.
Push *Cancel* to quit searching.


6.1.3. Connection Establishing

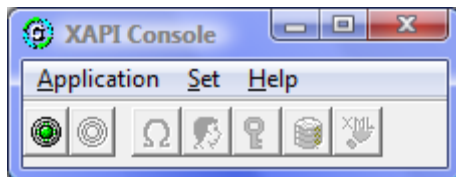
Having set up communication, try to establish connection with the 2N[®] XAPI server.

Push *Connect* . If the 2N[®] XAPI server is active, all the other buttons will become active. You need not push the *Connect*  button if you have opened connection using the automatic XAPI server search.




6.1.4. Connection Closing

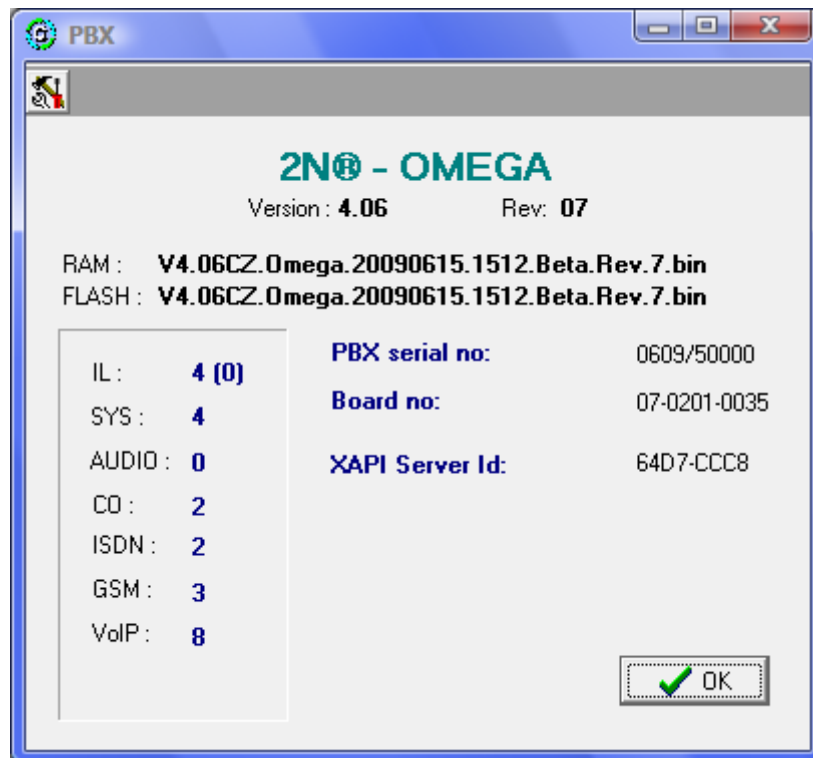
Push *Disconnect*  to close the 2N[®] XAPI server connection.



6.1.5. PBX Module

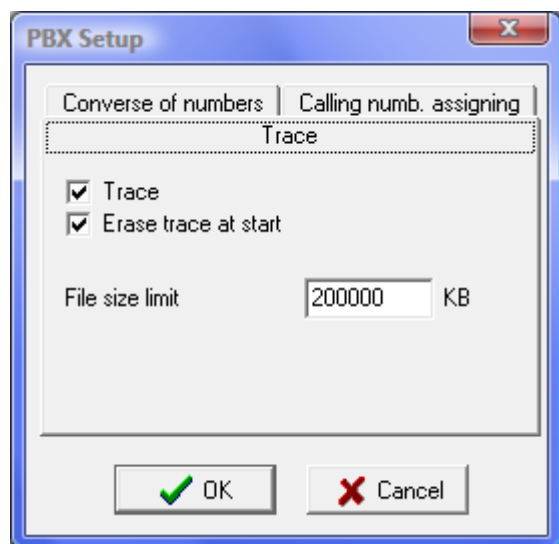
Push *PBX Module*  to access the PBX module.

A window gets displayed providing information on the PBX connected. It includes the PBX serial number, PBX firmware type, available boards, 2N® XAPI server ID (one of the XAPI server licence adding options).



Push *Setup*  to set the PBX module parameters.

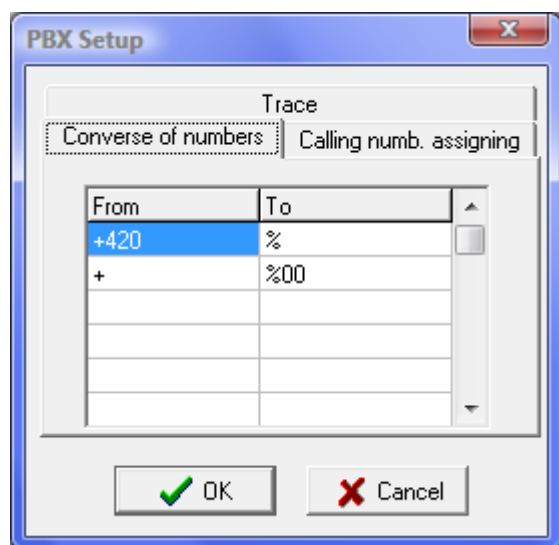
6.1.5.1. Trace



Enable/set the following:

- **Trace** – enable the creation of the PBX module LOG communication.
- **Erase trace at start** – enable the deletion of the PBX module LOG communication upon the XAPI server start.
- **File size limit** – limit the LOG file size (*.trc). When the limit is exceeded, the oldest data are overwritten.

6.1.5.2. Number Conversion

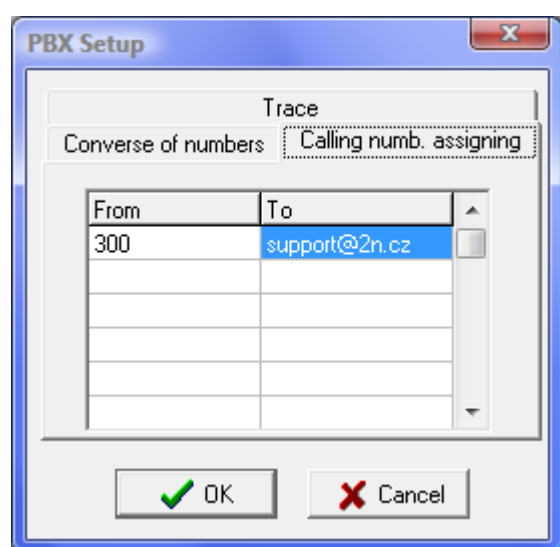


This menu helps you normalise received/called numbers to/from the current applications (PBX Assistant, TAPI driver, etc.). Remember to set this parameter to prevent dialling unsupported local and international calls to some outgoing connection types (ISDN, e.g.).

Set the following:

- **From** – original number format before normalising.
- **To** – resultant number format after normalising (% is always the first character in the format).

6.1.5.3. Called Number Assigning




This parameter helps you see the called party identification in the application (PBX Assistant). This function is suitable for reception desks that are shared by multiple PBX users.

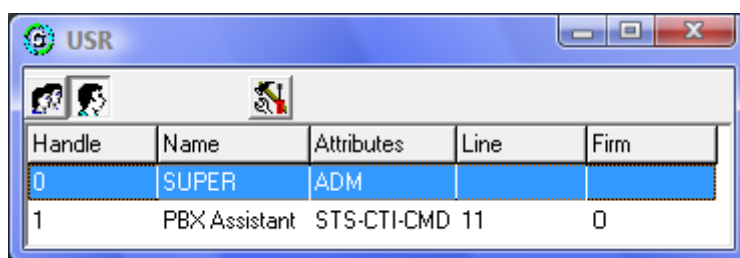
Set the following:

- **From** – DISA or DDI for ISDN connections.
- **To** – name to be displayed in the application.

6.1.6. User Module

Push *USR Module*  to enter the User module.


The User module is designed for creation of application users (clients) and displays the current connections.

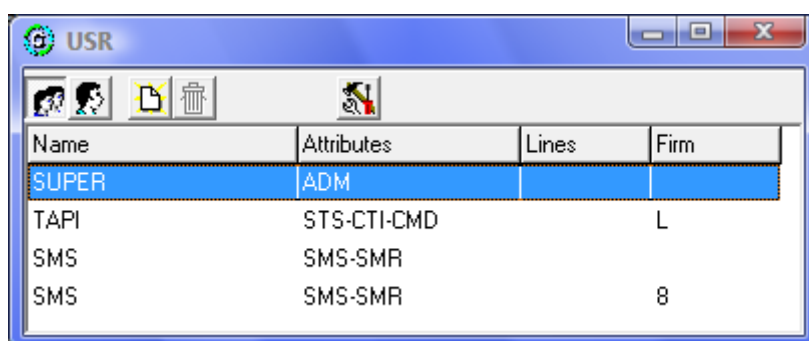


Handle	Name	Attributes	Line	Firm
0	SUPER	ADM		
1	PBX Assistant	STS-CTI-CMD 11		0

Meanings of USR parameters:


- **Handle** – number of the currently active connection.
- **Name** – user name.
- **Attributes** – user rights.
- **Line** – line used by the currently active connection.
- **Firm** – FirmCode (application) used for the currently active connection.

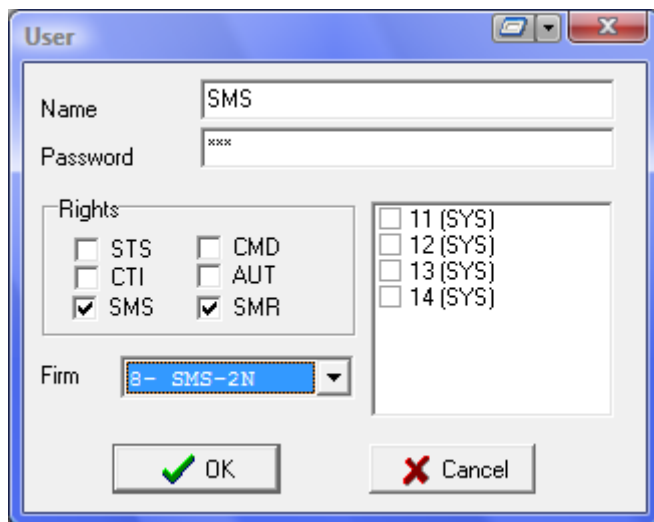
Push  to enter the user settings.



Name	Attributes	Lines	Firm
SUPER	ADM		
TAPI	STS-CTI-CMD		L
SMS	SMS-SMR		
SMS	SMS-SMR		8

6.1.6.1. User Creation

Push  to enter the user creating menu.




The image shows a 'User' configuration dialog box. It has a title bar with a blue gradient and standard window controls. The main area contains several fields and checkboxes. The 'Name' field is set to 'SMS'. The 'Password' field is masked with 'xxxx'. Under the 'Rights' section, there are two columns of checkboxes: the first column has 'STS', 'CTI', 'SMS' (checked), and 'SMR' (checked); the second column has 'CMD', 'AUT', and 'SYS' (checked). To the right of these is a list box containing '11 (SYS)', '12 (SYS)', '13 (SYS)', and '14 (SYS)'. At the bottom left, the 'Firm' dropdown menu is set to '8- SMS-2N'. At the bottom right, there are 'OK' and 'Cancel' buttons with green and red checkmarks respectively.

Enter the following:

- **Name** – user name.
- **Password** – user login password.
- **Rights** – user rights (the user rights included in the application licence may only be assigned).
- **Line** – line used by the currently active connection.
- **Firm** – FirmCode (application) used for the currently active connection.

Push *OK* for confirmation, or *Cancel* to clear the setting.

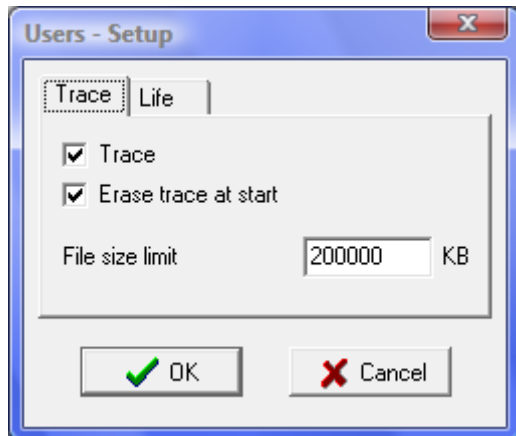
6.1.6.2. User Deletion

Select a user with the right-hand mouse button and push the  icon to delete the user.

6.1.6.3. User Module Setting

Push **Setup**  to set the User module parameters.

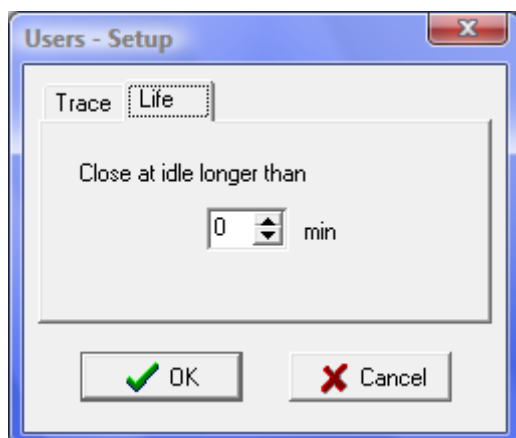
Trace



Enable/set the following:

- **Trace** – enable the creation of the User module LOG communication.
- **Erase trace at start** – enable the deletion of the User module LOG communication upon the XAPI server start.
- **File size limit** – limit the LOG file size (*.trc). When the limit is exceeded, the oldest data are overwritten.

Life



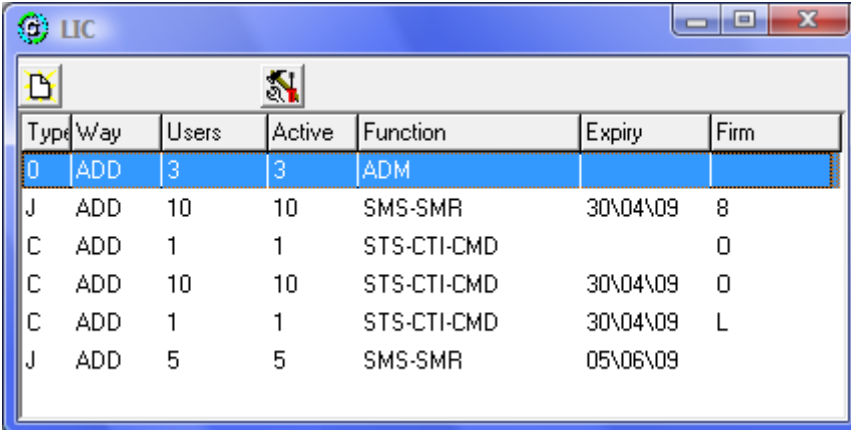
Enter the following:

- **Close at idle longer than** – define the idle time after which the application shall be closed. Most applications are provided with protection and thus retain the *0min = no forced disconnection* setting unless provided otherwise.

6.1.7. Licence Module

Push *Licence Module*  to enter the Licence module.

The Licence module is designed for adding and monitoring of application licences.



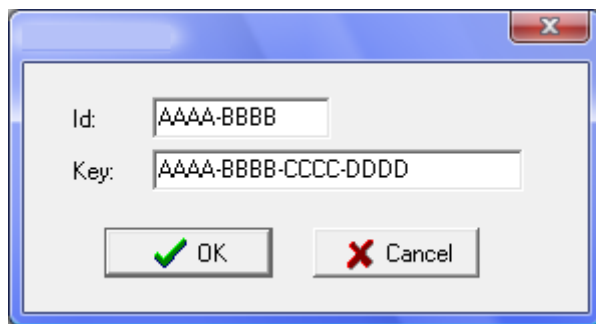
Type	Way	Users	Active	Function	Expiry	Firm
0	ADD	3	3	ADM		
J	ADD	10	10	SMS-SMR	30\04\09	8
C	ADD	1	1	STS-CTI-CMD		0
C	ADD	10	10	STS-CTI-CMD	30\04\09	0
C	ADD	1	1	STS-CTI-CMD	30\04\09	L
J	ADD	5	5	SMS-SMR	05\06\09	

Meanings of Licence module parameters:

- **Type** – licence type.
- **Way** – ADD is used as a rule.
- **Users** – maximum count of users (clients).
- **Active** – maximum count of users (clients) of simultaneously active connections.
- **Function** - user rights.
- **Expiry** – licence validity limit.
- **Firm** – FirmCode (application) used for the currently active connection.

6.1.7.1. Licence Adding

Push *New licence*  to add a licence.



A dialog box with a blue title bar and a close button (X) in the top right corner. It contains two text input fields. The first field is labeled 'Id:' and contains the text 'AAAA-BBBB'. The second field is labeled 'Key:' and contains the text 'AAAA-BBBB-CCCC-DDDD'. Below the fields are two buttons: 'OK' with a green checkmark icon and 'Cancel' with a red X icon.

Enter the following:

- **Id** – first licence part.
- **Key** – second licence part.

Push *OK* to add the licence.

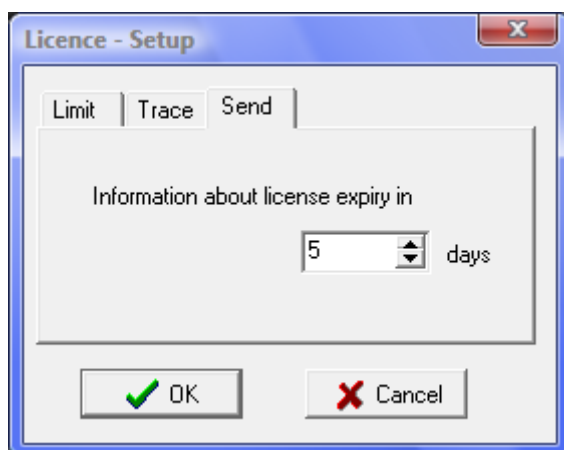
A licence is granted by the manufacturer with the following data:

- PBX serial number – information on the PBX;
- PBX main board serial number – information on the PBX;
- Application type;
- Count of users (clients);
- Name of the licence holder.

6.1.7.2. Licence Module Setting

Push *Setup*  to set the Licence module parameters.

Send licence info

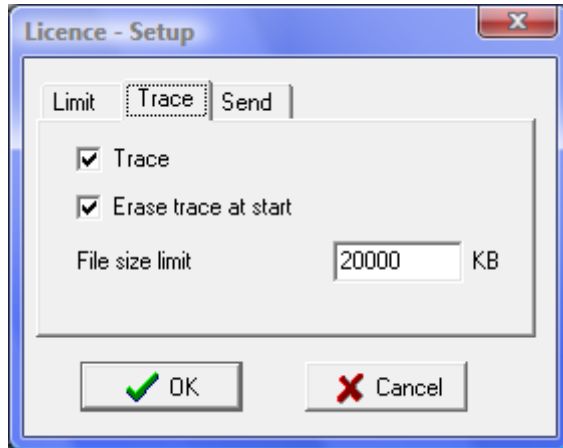


A dialog box titled 'Licence - Setup' with a blue title bar and a close button (X) in the top right corner. It has three tabs: 'Limit', 'Trace', and 'Send', with 'Send' being the active tab. The main area contains the text 'Information about license expiry in' followed by a spinner box showing the number '5' and the word 'days'. At the bottom are 'OK' and 'Cancel' buttons with green checkmark and red X icons respectively.

Enter the following:

- **Licence expiry in** – number of days till the licence expiry. It is displayed whenever the application start is executed or attempted.

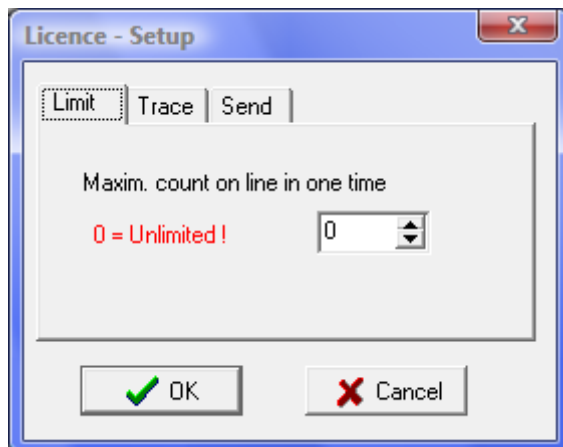
Trace



Enable/set the following:

- **Trace** – enable the creation of the Licence module LOG communication.
- **Erase trace at start** – enable the deletion of the Licence module LOG communication upon the XAPI server start.
- **File size limit** – limit the LOG file size (*.trc). When the limit is exceeded, the oldest data are overwritten.

Limit



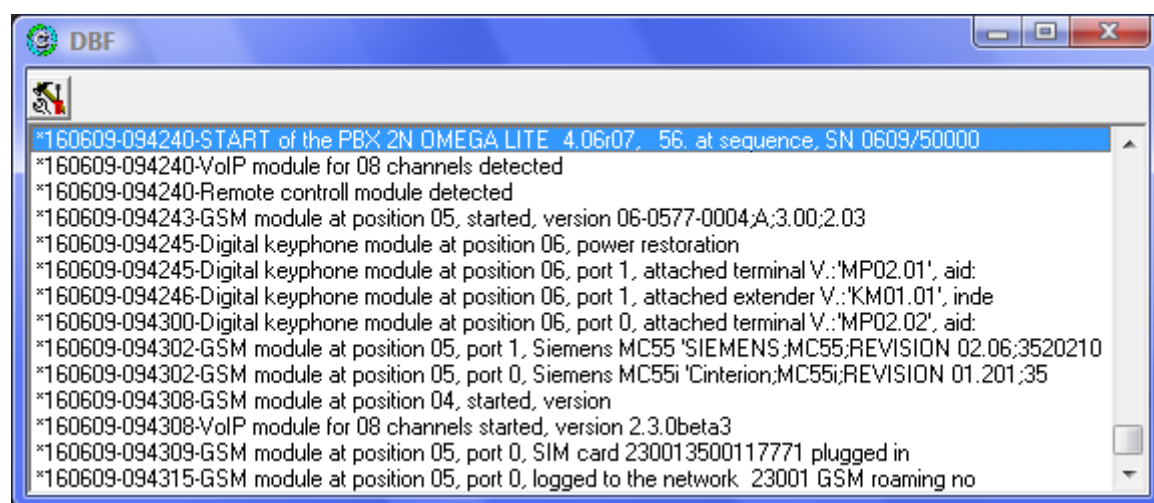
Enter the following:

- **Max. currently open connections** – count of application connections active on a line at the same time.

6.1.8. Database Module

Push *Database Module*  to enter the Database module.

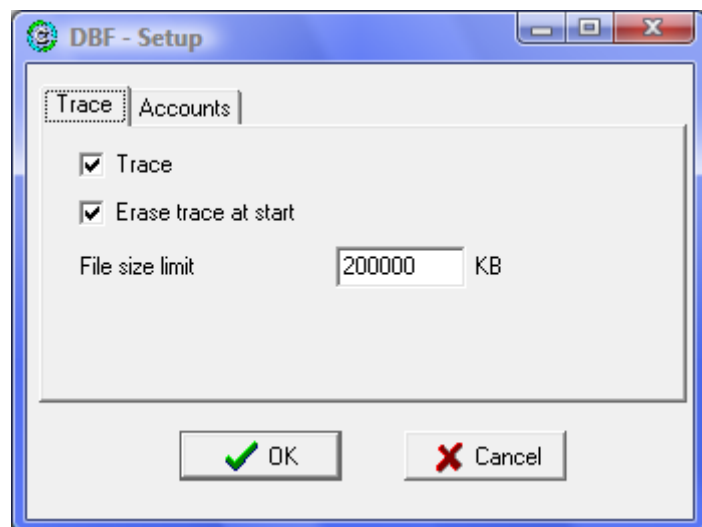
The database module is designed for monitoring and saving of accounting and system data.



6.1.8.1. Database Module Setting

Push *Setup*  to set the Database module parameters.

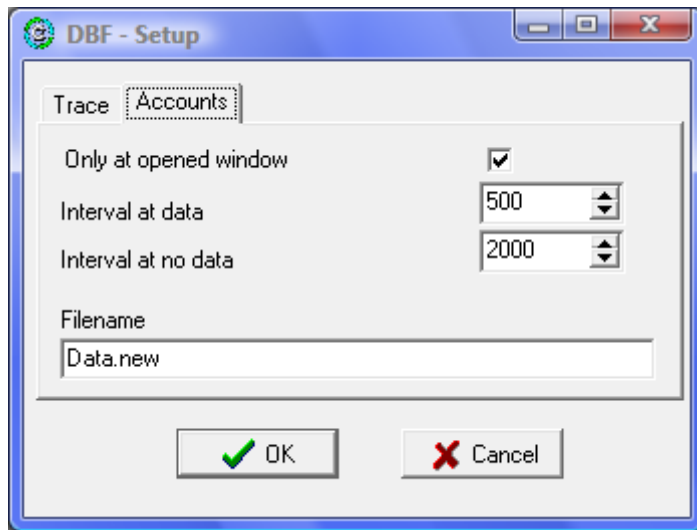
Trace



Enable/set the following:

- **Trace** – enable the creation of the Database module LOG communication.
- **Erase trace at start** – enable the deletion of the Database module LOG communication upon the XAPI server start.
- **File size limit** – limit the LOG file size (*.trc). When the limit is exceeded, the oldest data are overwritten.

Accounts



Enable/set the following:

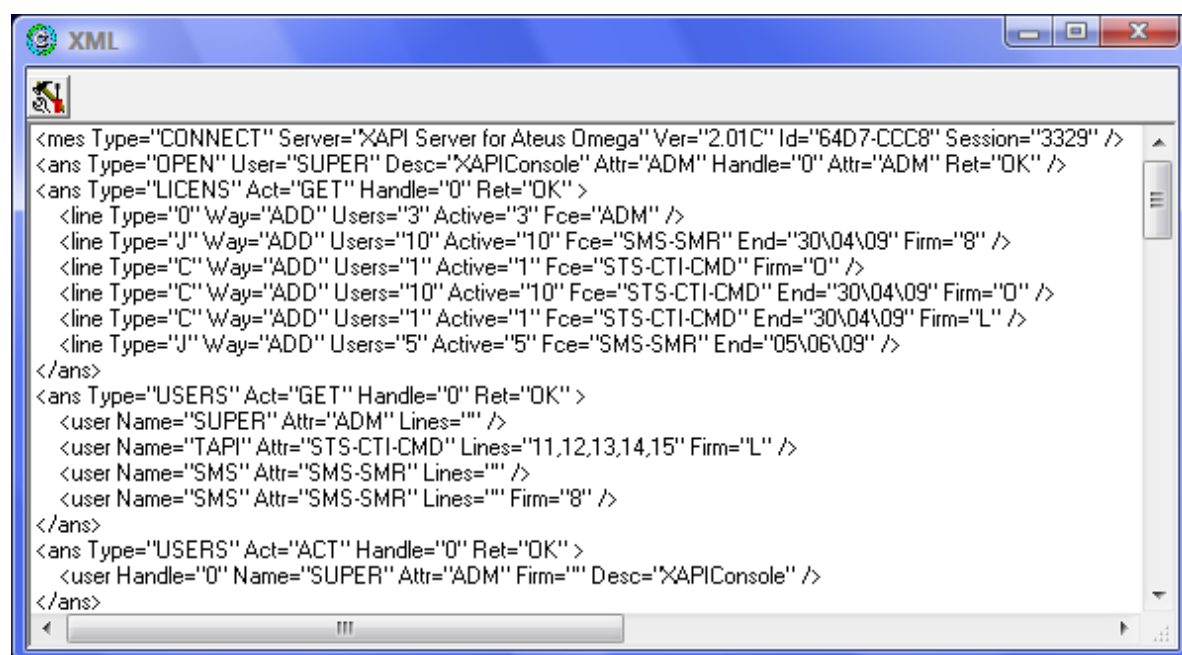
- **At open window only** – enable data downloading either *At open window only* or *Always*.
- **Interval at data** – data query interval [ms] if earlier data are available.
- **Interval at no data** – data query interval [ms] if no earlier data were available in the last query.
- **Filename** – filename for saving downloads. The storage is on the XAPI console installation site.

Push *OK* for confirmation.

6.1.9. XML Module

Push *XML Module*  to enter the XML module.

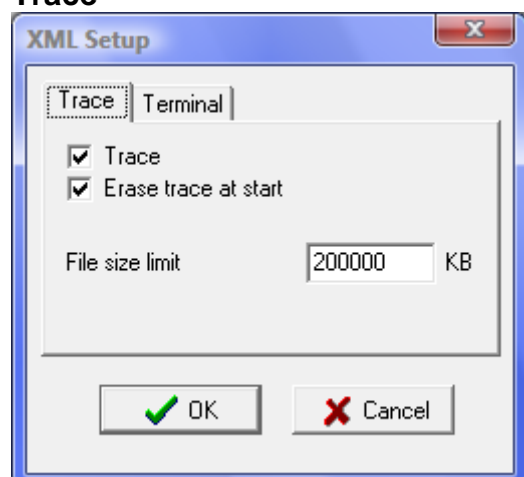
The XML module is designed for monitoring of XML communication with the open applications.



6.1.9.1. XML Module Setting

Push *Setup*  to set the XML module parameters.

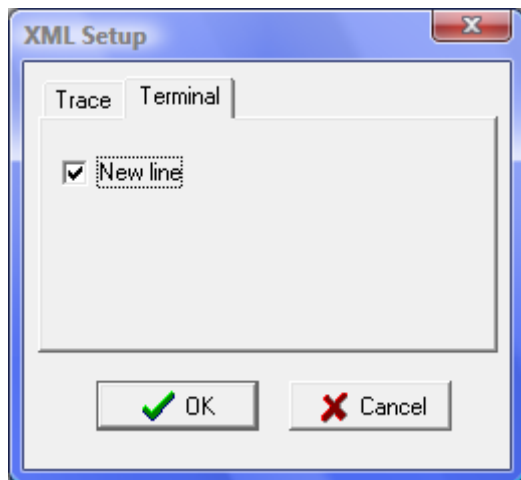
Trace



Enable/set the following:

- **Trace** – enable the creation of the XML module LOG communication.
- **Erase trace at start** – enable the deletion of the XML module LOG communication upon the XAPI server start.
- **File size limit** – limit the LOG file size (*.trc). When the limit is exceeded, the oldest data are overwritten.

Terminal



Enable the following:

- **New line** – enable closing of a line and proceeding to the next one for XAPI console connections (PuTTY, e.g.). This parameter is not used in normal operation.

***The manufacturer reserves the right to make such modifications in the documentation that will result in improvement of the product qualities.
Use the product in compliance with the Instructions for Use and for the purposes mentioned therein only.
When expired, the product or parts thereof should be disposed of in accordance with the applicable environmental control regulations.***



2N TELEKOMUNIKACE a.s.

Modřanská 621, 143 01 Prague 4, Czech Republic
Tel.: +420 261 301 500, Fax: +420 261 301 599
E-mail: sales@2n.cz
Web: www.2n.cz